Serious men make good clowns, the Hopi say, for in their language the word for "clowning" means "to make a point."

Kenneth Lincoln (1993)

Modular Sound Synthesis
A Learner's Glossary

This "Learner's" glossary is based on the idea that these definitions can enhance exploration of interrelated concepts and principles. Many entries provide more information than is necessary to frame a simple definition. Only someone who already knows a great deal about a field benefits from the terse definitions typical of so many glossaries, as context is (understandably) lacking. This glossary is intended to be more than a quick means of looking up individual terms–it's meant to be a more-global learning experience. Some entries constitute the only discussion of their topic within the extended publication. Accordingly, this glossary isn't merely a rehash of other text–some of the information here stands alone. These definitions are not written in an intentionally "learned" academic style, but neither have they been "dumbed down." Accordingly, it may become necessary to look up a word in a standard dictionary every now and then. In many instances, less familiar words and technical concepts within an entry are explained immediately within parentheses. This intentional redundancy might seem tiresome, or even condescending to those who have prior experience in the field, but redundancy aids newcomers immeasurably. This Learner's Glossary, and entire publication are skewed toward the needs of the intelligent neophyte, especially newcomers who are diligent and willing to get to work. Our entire publication is dedicated to adults of all ages–those who have learned the value and necessity of teaching themselves–by a couple of authors who were "old souls" at a tender age.

You are urged to follow the directives to "see" or "compare with" (cf.) related glossary entries to enhance learning. Expertise springs from creating a web of associations, and the orientation of this glossary facilitates that mission. The focus of the Learner's Glossary is broad, but not as deep as other parts of the text. This focus is geared toward longer-term exploration of sound synthesis, while pointing toward a number of allied fields such as acoustics, audio engineering, etc. You could read this glossary like a book, from cover to cover. The plot would admittedly be weak, but acquainting yourself with this cast of characters might be invaluable. Exploring this glossary sharpens your understanding of technical terms, and accurate vocabulary is the basis of mastering any technical language–such as modular sound synthesis.

This glossary conforms to Système International d'Unités (SI, or International System of Units) for grouping of numerals. For example, the numerals for one hundred ten thousand volts appear in the "AC" entry below, and this number falls under the SI category that has five or more digits. SI convention calls for such numbers to be depicted using three digit groups separated by a space rather than a comma. One hundred ten thousand volts is shown: 110 000 volts (110 kilovolts, or 110 kV). This is adhered to on both sides of the decimal point. One microvolt (1.0 μV) would be shown: 0.000 001 volts, which is one millionth of a volt, shown as (1×10^{-6} V) in scientific notation. Four digit SI numbers are depicted without a space or a comma. For example, one thousand hertz (Hz), formerly cycles per second (cps), is shown as: 1000 Hz (1 kHz, or 1 kilohertz), the frequency of a common reference tone in audio engineering and acoustics. SI conventions are followed regarding grouping of numerals because confusion arises otherwise, as many countries recognize the comma as a decimal point. See Douglas-Young (1987) for exhaustive expositions of SI units and the metric system, the basis for most SI units. Happy glossing!

abscissa • The axis for "x," or the horizontal value of a point on a two dimensional system of (x,y) Cartesian coordinates. See Cartesian coordinate. That is, the distance from a given point on a two dimensional graph to the vertical (y) axis, measured parallel to the horizontal (x) axis. This "x" value of the abscissa is positive (+) when the point is right of the central bisecting vertical (y) axis, or negative (–) when the point is left of the central vertical (y) axis. A specific abscissa (x value) and ordinate (y value) form a coordinate pair (x,y) used to locate any point in planar Cartesian space, two dimensions arrayed on a single plane. Cf. ordinate. • The (x) axis is properly known as the axis of abscissas, not simply as "the abscissa," as there are an infinite number of distances from the (y) axis, measured parallel to the (x) axis, and therefore an infinite number of abscissa values.

absolute value • Magnitude, or size of a number without regard to its plus or minus (±) sign, or in signal terms, its polarity. Absolute value is shown by placing a number between two short vertical lines: I – 49 I (negative forty nine) has an absolute value of forty nine, as do 49, + 49, and I + 49 I.

AC • Alternating current (AC) is a low frequency bipolar (±) nominally sinusoidal signal used to power electrical devices. Alternating current (AC) reverses its direction, or polarity periodically as it flows on an electrical conductor such as a copper wire. AC frequency for this periodic polarity reversal (60 Hz USA, 50 Hz EU) is strictly regulated by government agencies, because this frequency is widely used as a timing reference, e.g. to control electrical motor speed. Household AC amplitudes fluctuate as the power grid (electricity system) conditions change but are nominally 120 and 240 volts in USA. AC brownout is caused by power grid problems that result in a "sag" of household voltage levels to lower than nominal levels, which may cause various electronic devices such as computers to malfunction. An AC surge is typically a brief event that causes amplitudes that are much higher than normal, which may destroy electronic equipment in extreme cases. • AC is transmitted over long distances with less energy loss than direct current (DC), making AC preferable in a centralized power system. AC levels of 110 000 volts (110 kV) and higher are transmitted using long distance "high tension" (high voltage) power lines. Cf. DC. See sinusoidal. See (polarity) bipolar. Cf. (polarity) unipolar.

acoustic • Relating to musical instruments or other sound sources that directly generate and propagate (transmit in space) sound waves, without recourse to electronic means of generation and/or transmission. • Relating to performers or ensembles who play such instruments or sing without aid of amplification. See acoustics.

acoustics • The science and technology of the production, propagation, and perception of sound. The objective study of wave behavior in the audio frequency (20 Hz–20 kHz) part of the total spectrum of possible wave frequencies. Acoustically borne sound waves are not part of the electromagnetic (EM) spectrum, as sound is a longitudinal mechanical wave that requires a medium such as air through which it can propagate (travel). See longitudinal. Cf. transverse. Waves in the EM spectrum require no medium for propagation, e.g. light propagating through the vacuum of outer space. • The study of human perception of sound, i.e. its subjective aspects, is known as psychoacoustics. The study of sound within enclosed spaces (e.g. recording studios and other buildings) is known as architectural acoustics. The study of signals that can be converted into sound using electronic devices that eventuate in transducers (e.g. loudspeakers) is known as electroacoustics. See transducer.

acronym • In most cases, a pronounceable artificially constructed word comprising capital letters, each of which is taken from the first and/or other letters selected from those found within several related words. An acronym embodies or identifies an idea, device, linguistic phrase, technical standard, mnemonic (memory aid), protocol, or organization's name. For example: MIDI (pronounced mid´ee, Musical Instrument Digital Interface); SMPTE (pronounced simp´tee, Society of Motion Picture & Television Engineers); and CD-ROM (Compact Disc-Read Only Memory) are acronyms. See mnemonic.

ACT • Acronym coined by the authors that facilitates categorization of, and differentiation among audio, control, and timing (ACT) signals in a modular sound synthesis system. ACT is spoken like the word "act" and stands for audio, control, timing. The ACT dictum (rule) states that any module's output, and therefore any signal in the system functions as an audio, control, and/or timing signal based solely on the type of ACT input(s) to which that signal is connected. For example, an oscillator is an audio (A) signal generator only when connected to an audio input or bus that terminates at an audio monitor. An oscillator may certainly be categorized a priori as a periodic waveform generator based on its inherent features or measurements of its output signal characteristics. However, prior to some connection of its signal output to an audio (A) input, an oscillator should not be categorized as an "audio (A) signal generator," regardless of its inherent frequency span, digital signal processing (DSP) bandwidth, frequency resolution, or any output signal characteristic. Accordingly, the functions of modules and signals remain undefined prior to their connection to specific ACT inputs. The ACT dictum: the functions of a signal are not determined a priori by its bandwidth or signal characteristics, but by the functions of inputs to which it is connected. • Acting on the ACT principle is enabling. ACT encourages divergent use of modules by precluding a priori, i.e. output-oriented categorizations of modules that substantially limit the myriad ways modules and signals might function when categorized appropriately, i.e. solely due to their connections to ACT inputs. Categorization based on convergent thinking tends to cause modules and signals to be used stereotypically. The validity and utility of ACT to enhance creative use of modules has been proved by the authors perennially in the classroom and elsewhere. See a priori. Cf. a posteriori. See bandwidth.

ADC • Analog to digital converter. See (converter) ADC.

ADSR • Acronym for attack (A), decay (D), sustain (S), and release (R) segments of the aperiodic signal produced by the popular ADSR envelope generator (EG) module. See signal, aperiodic. Each letter stands for an individual distinct signal element, or segment. These segments are generated serially in the order A-D-S-R when the EG module is "triggered" or "gated" by a signal connected to its gate input. A (attack), D (decay), and R (release) segments each have an individually programmable time, and typically involve a smooth, continuous transition between two discrete levels during that segment's user-programmed time. The S (sustain) segment has no programmable time. It has a programmable level ranging from zero (0) to a positive (+) maximum level equivalent to full scale (FS), i.e. the largest possible output level the ADSR module can generate. Timing of the S segment typically depends on the gating signal connected to EG gate input. • Unlike an oscillator or periodic function generator, an envelope generator (EG) nominally produces an aperiodic signal–one that is not automatically produced and endlessly repeated. Therefore, an ADSR module or any other envelope generator (EG) must be "signaled," or told when to initiate elements of its output signal. This is done by connecting a signal to the envelope generator's (EG) timing input, known variously as a gate or trigger input. The "on" gating condition starts the first segment ("A") produced by the EG module. The "off" gating condition starts the last segment ("R") produced. Typically, the sustain level (segment "S") is maintained so long as the gating signal remains at its "on" level. See input, timing. See input, gate. See threshold detector. See gate signal. • The idea of the ADSR envelope generator was conceived by Vladimir Ussachevsky of Columbia-Princeton Electronic Music Center, and first developed as a hardware module by Bob Moog as part of his earliest 900 Series Modular systems. "ADSR" is spoken by saying each letter (A-D-S-R) serially, an acronym that is not a pronounceable word. "ADSR" is also used to designate the particular EG module that produces four such segments, as well as to describe that segmented signal itself. See full scale. See generator, envelope. See EG segment.

address, computer • Unique numerically encoded location in a memory matrix or array, or some other part of a digital computer.

address, SMPTE time code (TC) • SMPTE time code is a synchronization standard promulgated by the Society of Motion Picture and Television Engineers (SMPTE, pronounced simp´tee). Temporal (timing) data are coded in binary coded decimal (BCD) digits in hours (H), minutes (M), seconds (S), and frames (F). Each time subdivision is displayed as a two digit decimal number HH:MM:SS:FF starting from, e.g. 00:00:00:00 and running through 23:59:59:23 for a frame rate of 24 frames per second (fps). A choice among several time code subdivisions of each second, i.e. frames per second (fps) is available for American film (24 fps), European film (25 fps), NTSC color television (29.97 fps), and older USA black and white television (30 fps). That is, unique time codes are displayed per each film or video frame as decimal number addresses, with two digit groups representing hours (0–23), minutes (0–59), seconds (0–59), and frames (0-23), e.g. in a 24 frame rate environment. These HH:MM:SS:FF numbers are displayed simultaneously on a video monitor arrayed from left to right, e.g. for a 24 Hz frame rate displayed as 11:56:08:00 which represents an address of 11 hours, 56 minutes, 8 seconds, and the "zeroeth" frame (the first of twenty four possible frames in this example) in an ongoing time code data stream. The frame numbered "23" is the twenty fourth frame when a frame rate of 24 is selected. Following frame 23, the limit of twenty four frames is reached, the frames count then resets to 0, at which time the "seconds" number is incremented by 1, and so forth to eventually include incrementing of minutes and hours. • SMPTE time code is nominally "temporal" (time-related) data, but actually provides a means of marking discrete locations on moving media recordings (analog tape) or digital audio computer files. A succession of such uniquely numbered time code addresses that include hour, minute, second, and frame numbers are produced serially by an electronic time code generator. These data are encoded as a bi-phase mark modulated signal, a form of binary signal encoding. This time code signal may be recorded, like any audio signal, thereby marking unique "time-locations" on a moving track of analog magnetic audio tape. Or, time code may function similarly on a virtual track of a hard disk recording system. • Such SMPTE LTC (longitudinal time code) features 80 bits per computer "word" (unique LTC address), and each 80 bit word comprises a single unique frame. Selection of, e.g. a 30 Hz frame rate produces an easily recordable audio-range signal whose quasi-periodic "frequency" warbles somewhere between 1200 Hz (all zeroes), and 2400 Hz (all ones). That is, 30 frames per second times 80 bits per frame yields a nominal 2.4 kHz upper frequency limit (given repeated address frames that have only ones). In practice, LTC 80 bit computer words (Time Code addresses) are not populated solely by either all zeroes or all ones, thus the "quasi-periodic" nature of the LTC signal. • Time code tracks on different audio tapes previously recorded, or "striped" with SMPTE time code can be synchronized using an electronic synchronizer governed by a master timing reference, a clock known as house sync (sync, or synch is pronounced like kitchen "sink.") When recorded time code tracks are synchronized, or "in sync," the accompanying recorded program material (music, sound effects, etc.) on associated tracks will also be synchronized, if the system has been used properly. • Such longitudinal time code (LTC) is casually called "SMPTE" (pronounced simp´tee), which is technically a misnomer. SMPTE is the acronym for the Society of Motion Picture & Television Engineers, an organization that maintains various technical standards, among which are those for longitudinal time code (LTC, pronounced lit´see or "L-T-C") and vertical interval (video-based) time code (VITC, pronounced vit´see). • LTC or VITC (time code) is expressed in temporal units (hours, minutes, seconds, frames), but these putative "times" do not necessarily represent or correspond to an actual time of day at any particular locale. SMPTE time code is typically used in a relative, rather than absolute temporal sense. (Some recording studios do attempt to maintain a correspondence between local "real," or "absolute" time, and SMPTE time code (LTC or VITC), thereby providing a record of the local time at which a recording is made in that studio.) Time code addresses recorded on different media (e.g. analog magnetic tapes, video tapes) at various studios that are geographically dispersed worldwide likely will have different "local times" represented by LTC and/or VITC time code streams. That is, it's possible to start the time code "clock" at any "time" for a particular recording. Disparate time code streams must then be synchronized by programming offsets in hours, minutes, seconds, and frames between such streams. So, the actual time of day, or indeed even the date at which a time code stream is recorded is essentially irrelevant. • Temporal (timing) resolution of SMPTE time code far exceeds the resolution implied by any selected frame rate. On the face of it, 24 or 30 frames per second (fps) implies a rather coarse temporal resolution between adjacent frames, or time code addresses, of one twenty-fourth (1/24 sec) or one thirtieth (1/30 sec) of a second respectively. SMPTE time code system temporal resolution is greatly improved by inclusion of a binary encoded sync word common to the bit structure of all time code addresses generated. This is part of the 80 bit LTC frame structure and is part of the computer word structure that is common to all frames. This sync word allows temporal resolving circuitry, e.g. a phase lock loop (PLL) and/or associated computer algorithms in the time code synchronizer to align time code streams with far better temporal resolution than any selected frame rate such as 24 or 30 fps nominally appears to offer. The number of bits in a SMPTE time code "word," or frame (LTC 80 bits, VITC 90 bits) does not determine overall temporal resolution, i.e. the "fineness" of possible subdivisions of time by the system. (An 80 or 90 bit temporal resolution per second in this scenario would provide timing resolution that exceeds comprehension, to say nothing of practical needs for synchronization!) It is the sync word with associated feedback & compare algorithms that provide increased temporal resolution that far exceeds any selected frame rate–not the number of bits in a LTC or VITC frame. See (measurement) resolution. • Digital word clock may supplant or supplement SMPTE time code(s) in many digital audio workstation (DAW) environments.

after touch (aftertouch) • So-called "continuous" digital data, a description that flirts with being an oxymoron. See oxymoron. (Digital data are expressed using a set of constrained discrete magnitudes, rather than continuous (analog) signal levels that would otherwise remain free to take on any value, subject to the limitations of available measurement tools.) Properly speaking, after touch might be described as a continuously ongoing digital data stream. Such data can be transmitted using the Musical Instrument Digital Interface (MIDI) protocol. In this context, after touch typically represents the variable amount of force applied from moment to moment to one or more keys depressed on a MIDI musical instrument keyboard. After touch data are sent after transmission of MIDI note on/velocity command(s) that initially cause note(s) to sound, hence the designation "after" touch. MIDI after touch is of two types: (1) channel, a single composite value that represents the total force applied to all keys depressed without differentiating among keys; or (2) polyphonic, where a separate force value is transmitted per each key depressed. Polyphonic after touch necessarily requires more digital signal processing (DSP), i.e. increases the burden on available bandwidth for transmitting data. Obviously, after touch is available only on a MIDI keyboard that provides the physical apparatus to actually sense application of force on the keyboard. MIDI always provides the means to transmit after touch data but can do so only if the keyboard in use actually produces such data. • MIDI after touch has erroneously been called "pressure," but pressure is a measurement of force exerted per unit area ($p = F/A$ where "A" is surface area, and "F" is force.) Air pressure, e.g. @ 30 pounds per square inch (psi) in automobile tires is an example. Force is the appropriate term to describe MIDI after touch. Force is the amount of energy expended due to a physical action or movement. After touch data represent the changing level of force expended over time while key(s) are depressed, whether force is exerted over a large surface area of the actual keys depressed, or not. For instance, force might remain at the same value whether a large human finger or a thin pencil were used to depress key(s). Surface area (A) on the keys depressed plays no role in MIDI after touch, hence the use of the term "pressure" to describe after touch is dubious at best. But it is equally doubtful that the more appropriate term force will be adopted, even should the MIDI protocol be updated. Why? See qwerty principle. • In historical terms, "after touch" is a capability of specific theater (pipe) organs, e.g. "The Mighty Wurlitzer," in which pressing key(s) to a (single) deeper level actuates any of a number of effects such as snare drum rolls, reiterated attacks on externally controlled instruments (e.g. idiophones, or mallet instruments), accented notes, etc.

aka • Non-pronounceable acronym or abbreviation for "also known as." Spoken by saying letters "a-k-a" serially or by speaking the words this acronym represents.

algebraic addition • Math operation whereby numbers of any sign (±) can be summed, e.g. negative numbers can be added to or subtracted from positive numbers. A negative number (e.g. –3) algebraically added to another number (e.g. 5) is placed within parentheses (–3) and is then effectively subtracted from the number: $5 + (-3) = 2$ (unlike signs do not cancel each other). A negative number (e.g. –3) subtracted from another number (e.g. 5) is also enclosed in parentheses (–3). Because negative signs cancel each other to become positive, the number is then added: $5 - (-3) = 5 + 3 = 8$. Algebraic addition is involved when two or more (input) waveforms are added (summed), in which case the instantaneous plus or minus (±) polarities of all added waveforms from moment to moment are taken into consideration in order to yield the summed waveform (output) polarity and amplitude from moment to moment. This constitutes constructive or destructive "interference" of the waves with one another, and may occur in the air, electronic circuitry, or algorithms for virtual systems. See interference, constructive. See interference, destructive. See polarity.

algorithm • Computer code, virtual (mathematical) procedure, or process with a finite series of steps that executes a specific task or solves a problem expected to occur frequently. • Term chosen by the Yamaha company for a "patch," or interconnection of linear frequency modulation (FM) operators, in their early popular line of FM synthesizers, starting with models DX7, DX9, etc.

alias • Sampled partial (sine wave) with an erroneous frequency caused by using a sample rate (SR) whose frequency is too low (slow) to accurately capture and represent that partial's actual frequency upon playback. See aliasing.

aliasing • Sampling anomaly, aka foldover that causes the frequency of a sampled partial (sine wave) to be changed to exhibit a spurious value, different from its frequency prior to sampling. • Sampling any partial (sine wave) whose frequency is higher than one half (½) the frequency of the sample rate (SR) causes that sampled partial to have an alias frequency. The simplified formula when this is the case is: sample rate (SR) – sampled frequency = alias frequency. Given a sample rate (SR) of 44 100 Hz, a sampled partial whose frequency is 38 000 Hz will be falsely represented by a frequency of 6100 Hz. The partial's original frequency of 38 000 is too high to be accurately captured by sampling at a SR of 44 100. No frequency that is greater than one half (½) the sample rate (SR) can be captured without aliasing. The frequency that is one half (½) the sample rate (SR) is called the Nyquist, Nyquist frequency, or Nyquist limit, 22 050 Hz in this example. See frequency, Nyquist. • An alias, analogous to an assumed name intended to conceal true identity, has a counterfeit frequency that replaces the aliased partial's genuine frequency. This reveals that aliasing does not create more partials in the reconstructed analog final output than were originally present in the original input signal. Rather, aliasing causes misrepresentations of frequencies of those partials present in the original signal prior to sampling, specifically for those frequencies that exceed the Nyquist frequency. • After aliasing has occurred, the frequencies of aliased partials can't be corrected simply. Nor can aliased partials be removed by the digital to analog converter (DAC) and "brick wall" (steep slope low pass) filter that convert a digital audio file back into the analog signal that was sampled originally. Aliased partials whose counterfeit (new) frequencies fall below the Nyquist frequency are converted by the DAC and passed by the low pass filter as apparently legitimate, and therefore become part of the reconstructed analog output signal. That is, the DAC and final output smoothing filter cannot distinguish between alias frequencies, and frequencies of partials that have been sampled correctly. Aliasing can be prevented by placing a brick wall (very steep slope) filter in front of the analog to digital converter (ADC) input (at the beginning of the sampling process–as well as the required brick wall filter following the digital to analog converter (DAC)). See ADC. See DAC. See filter, brick wall. See alias.

alphanumeric • See symbols, alphanumeric.

AM • Amplitude modulation. The alteration of the amplitude (size) of a carrier signal according to the signal characteristics of a modulator signal. See (modulation) AM. See BAM. See tremolo. Cf. vibrato.

ambient • In the local or immediately surrounding area, after the Latin ambient "going around," from ambire. Usage implies volatility, or fluctuations of a measurable or discernible condition in the surrounding area, e.g. ambient barometric pressure, ambient air temperature, or ambient sound. • The ambient sound in an enclosed space, a component of overall room tone, may include noise from: heat, ventilation, & air conditioning (HVAC) systems; the audience; and other internally or externally produced noises that might interfere with a musical performance or lecture in such a venue. Room tone is also influenced by the resonances, or modes, i.e. standing waves that occur due to the dimensions and shape of the room, as well as the absorption characteristics of the surfaces of that particular enclosed space. See room tone.

ampere • SI unit of measurement (symbol: A) for the amount of electrical current that passes through a given point on an electrical conductor in time, often contracted to "amp." Named in honor of the French physicist André Marie Ampère (1775–1836), first to distinguish the rate of passage of current (amperage) as different from the driving force of electricity, known as voltage. In technical terms, one (1) ampere equals one (1) coulomb of electric charge that passes through a single circuit point per second. The abbreviation amp for ampere, should not be confused with "amp," the common abbreviation of "amplifier."

amplification • Process by which the power, amplitude, or level of a signal is increased or decreased by a gain factor expressed as the ratio of some output to input (O:I) measure of signal size or power. An increase, or output to input ratio greater than unity (1:1) is commonly called gain, while a ratio of output to input that is less than unity, or decrease is known as a loss. Technically, the term gain may be used in either case: positive gain for an increase, or negative gain for a decrease of power, amplitude, or level. Negative gain is usually referred to as a "loss," however. See amplifier. Cf. attenuation.

amplifier • Broadband processor that provides a fixed, but typically alterable amount of gain that changes the instantaneous levels, (average) amplitude, or power of the signal processed. Amplifier gain is represented by the ratio of output (O) magnitude to input (I) magnitude (O:I). A 2:1 amp has a positive gain that doubles the amplitude or (in some cases) the power of the input signal, which is typically calibrated in voltage, decibels, Watts, or some numerical representation in a virtual system. Amplifier gain can be stated in decibels (dB), a logarithmic measure of the relative power of two signals, or relative power of a single signal compared to an established reference power. See (measurement unit) decibel. A 6 dB (decibel) increase is a doubling (2:1) of input signal amplitude, which is a quadrupling (4:1) of input signal power. This difference in ratios is due to the square relationship between signal amplitude and power. See relationship, square. Amplifier is often abbreviated as amp. See amplification. See multiplier. • The ideal high fidelity audio amplifier would have a flat frequency response. See frequency response, flat. (Guitar amplifiers are intentionally designed to "color" the signals they amplify, and therefore do not necessarily have a flat frequency response). That is, a high fidelity (recording studio) amplifier should not color the sound by significantly altering the processed signal's tonal balance–the relative strengths of low, mid, and high frequency components (partials). In practice, any amplifier can be designed to be only approximately flat over a limited bandwidth (span of frequencies), as indicated by the plus or minus (±) specification that enumerates, in decibels (dB), the largest deviation from a theoretical perfectly flat frequency response. This specification of deviation from perfect flatness in dB should include the span of frequencies covered, e.g. 20 Hz – 20 kHz. An amplifier with a nominally (nearly) flat frequency response changes amplitudes of all partials in the signal it processes proportionately, without significantly altering other signal characteristics, e.g. waveform or frequency. See frequency response, flat. See tonal balance.

amplifier, differential • A module, e.g. amplifier, that has a pair of input and/or output jacks that provide signals with opposite (±) polarities, i.e. both plus (+) and minus (–) has differential inputs and/or outputs. See jacks, differential input. See jacks, differential output. Cf. amplifier, inverting.

amplifier, inverting • Broadband processor with a gain that negates (–), or reverses the (±) polarity of the input signal. Positive (+) levels become negative (–), and negative (–)

levels become positive (+) when inverted. See relationship, reciprocal. See inverse. • A change of polarity is technically not a "phase" change, a common misnomer. A phase change occurs due to a temporal (time) anomaly such as the delay of one signal relative to another.

amplifier, negative gain • Broadband processor with a gain, e.g. (1:2) that is less than unity (1:1) gain. Negative gain, or loss decreases input signal amplitude. • A fixed attenuator, or switchable pad on a recording console acts like a negative gain amplifier. However, in analog terms, a pad is more likely an electronic circuit that reduces signal level by shunting (diverting) part of the processed signal through a resistor to electrical ground, thereby dissipating a percentage of that processed signal's power as heat. • In digital signal processing (DSP), dividing each element in the stream of numbers that represents signal levels by a selected constant, e.g. 2 causes a negative gain (1:2). This 1:2 ratio represents a loss, aka negative gain, and is equivalent to a 50% attenuation (reduction) of the input number stream that represents signal levels. See attenuation. See constant. Cf. amplifier, positive gain.

amplifier, positive gain • Broadband processor with a gain e.g. (2:1) that is greater than unity (1:1). Positive gain, or boost increases input signal amplitude or power. • In digital signal processing (DSP), multiplying each element in the stream of numbers that represents signal levels by an appropriate selected constant, e.g. 2 causes a positive gain (2:1). This 2:1 ratio represents a 100% "boost" (increase) of the input number stream that represents signal levels. See amplification. See constant. Cf. amplifier, negative gain.

amplifier, summing • Amplifier with unity (1:1) gain that algebraically adds (±) several input signals, but changes no input signal's original input level. • Inputs on a typical mixer are connected to a unity gain, or summing amplifier. Positive mixer gain is provided by a positive gain amplifier connected in series to the output of this input summing amplifier. • The external and internal control inputs on a synthesis module are typically connected to a summing node, i.e. a summing amplifier that algebraically adds (±) input signals without altering their individual levels or other signal characteristics. Therefore, for most sound synthesis modules, several control signals connected to a module's control inputs will sum (add to each other).

amplifier, unity gain • Broadband processor with a gain of one (1:1), or unity. A unity gain processor does not increase the level of the signal passing through it. For example, a unity gain voltage controlled amplifier (VCA) can change the level of a signal it processes, but does not provide positive gain, even at full scale (FS) output. See full scale. A unity gain VCA can only attenuate (decrease) an input signal's amplitude–it cannot increase it. Such a VCA is useful, and technically it is a multiplier (amplifier), even though this categorization may not make intuitive sense. (Multiplication by "1" remains multiplication.) • In terms of digital signal processing (DSP), voltage controlled amplifier VCA or multiplier output is the instantaneous products, i.e. the result of multiplication, of the two streams of numbers that represent signals at its carrier (C) and modulation (M) inputs from moment to moment. See multiplier. See amplifier, voltage controlled.

amplifier, voltage controlled • A voltage controlled amplifier (VCA) is an analog processor that changes the levels of the signal passing through it proportional to the signal(s) connected to its control input(s). • A VCA has two kinds of inputs, signal input and control input. VCA instantaneous signal output is the product that results from multiplying instantaneous levels present at its signal and control inputs. The output level of the processed signal can be changed dynamically by using a dynamic control signal, e.g. the output of an envelope generator (EG). See ADSR. • The classic analog VCA is a two quadrant multiplier, because it responds to a bipolar (±) signal at its signal input, but can output a signal only when its control input receives a positive (+) signal. See multiplier, two quadrant. (A four quadrant multiplier functions with bipolar (±) signals at both signal and control inputs. See multiplier, four quadrant.) Any signal connected to the control input on a two quadrant multiplier does not appear at that multiplier's signal output. Conceptually, only the processed signal connected to the signal input can pass through this processor, i.e. the carrier signal. • Virtual, i.e. digital signal processing (DSP) systems may not provide both types of multipliers or may fail to make a distinction between the two quadrant and four quadrant multiplier. See multiplier.

amplitude • Generally, a measure of the overall, or averaged (integrated) magnitude (size) of a varying signal or vibration with respect to zero (0), in contradistinction to any of that signal's individual instantaneous levels from moment to moment. • The classical definition of amplitude is that signal characteristic represented by the size of a bipolar (±) sine wave in the time domain, measured from its zero level to its largest positive (+) value on the vertical (y) axis, i.e. an ordinate value. See ordinate. This measurement is aka peak amplitude. Peak amplitude is equal to ½ of the peak to peak, or crest (+) to trough (–) amplitude of a sine wave. • Most signals of musical interest are complex waveforms, not sine waves. Therefore, other measures of amplitude may be more practical or have greater relevance to the sound designer or recording engineer, particularly because the commonplace audio engineering VU (volume unit) meter provides an average signal value, not the peak value. See amplitude, average. See amplitude, RMS (Root Mean Square). • The following are relationships among various measures of amplitude:

RMS = 0.707(peak) = 1.11(average)

peak = 1.414(RMS) = 1.57(average)

average = 0.637(peak) = 0.9(RMS)

peak to peak = 2.828(RMS)

RMS = peak to peak(0.3535)

• Amplitude is an objective signal characteristic, but the fluctuating (±) levels of an audio signal directly correlate with (±) changes of ambient sound pressure level (SPL) produced by that audio signal's monitored sound wave. SPL is perceived subjectively as loudness changes, a judgment of SPL or signal intensity, and can be measured objectively in decibels (dB). Amplitude is an objective signal characteristic that can be measured, e.g. in volts. Loudness is a subjective sonic attribute that can only be perceived or judged, but loudness can be related indirectly to objectively derived measurements in Phons or Sones. For certain, the terms amplitude and loudness are not synonyms, and should not be used interchangeably. Terms that describe signal characteristics such as amplitude, should not be automatically associated with sonic attributes such as loudness, despite the fact that standard signal-sound correlations do exist for some cases. See correlation. For example, the changing amplitude values of various control signals used in modular sound synthesis are not necessarily perceived as loudness changes. For instance, alteration of signal amplitude can produce changes of timbre, e.g. index of modulation in linear FM (frequency modulation), or frequency (depth of vibrato), etc. rather than loudness. See ACT. See equal loudness curves. See SPL (sound pressure level). See ambient.

amplitude, average • Signal characteristic that represents the size of a waveform, scaled on the (y) axis in the time domain, as measured by an instrument that has a direct current (DC) movement that includes a full-wave rectifier. A volume unit (VU) meter responds to the average, rather than the root mean square (RMS) value of amplitude. But, a VU meter is typically calibrated in RMS amplitude values, which causes inaccuracies when metering complex waveforms typical of audio engineering. Technically, an amplitude averaging display (e.g. VU meter) calibrated in RMS is accurate only for sine waves. Uh, but we use VU meters in audio engineering anyway. See rectifier. See amplitude, rms.

amplitude, effective • Another name for RMS amplitude, approximately 0.707 times peak amplitude. See amplitude, RMS.

amplitude, full scale • Full scale (FS) is the largest number used in a virtual, or digital signal processing (DSP) system to represent a signal's size, amplitude, level, magnitude, etc. Full scale (FS) is the largest ordinate (y) value that does not produce distortion, error, or a computer condition known as "overflow." See overflow.

amplitude, negative • A signal amplitude (magnitude) with a polarity that is opposite (–) that of one determined, or deemed to be positive (+). • Sideband synthesis technique(s) such as linear frequency modulation (FM) often produce partial(s) that have a negative amplitude. Negative amplitude does not mean "less than zero amplitude or no amplitude," just as a negative decibel (dB) level does not mean "less than zero power or no power."
Take note that zero decibels (0 dB) on a volume unit (VU) meter on a recording console indicates a relatively large signal (approximately 0.775 volts) in audio engineering, not a signal with zero amplitude or power. Typically, a "negative" (–) signal characteristic embodies the idea of measurement relative to some reference point deemed to be zero (0) or positive (+), rather than some absolute measurement or representation.

*amplitude, normalized • Result of a single digital signal processing (DSP) operation that causes the largest number (level) present in a digitized sound file to be increased to full scale (FS), which typically has a value of one (1.0) by design. All levels in the sound file increase proportionately when normalized. The process involves multiplication of all sample values by the particular factor that will increase the largest of those sample values to the full scale (system maximum) value. Normalization produces a digitized file whose potential signal to noise (S/N) ratio (with respect to quantization noise) is optimized–with no increase of distortion due to overflow. See overflow. Normalizing a digital signal does not "improve" the maximum signal to noise (S/N) ratio possible given a specific number of "bits" used to represent signal values. For example, the S/N ratio inherent to an eight (8) bit signal is approximately 48 dB (6dB per bit), and this S/N ratio cannot be exceeded due to normalization. Normalization simply brings the largest existing level of the signal to system full scale (FS) value, thereby increasing all other levels proportionately. In this example, normalizing any "under-recorded" signal would also raise the level of quantization error in that signal, and would not necessarily yield the maximum possible 48 dB signal to noise (S/N) ratio inherent to an eight (8) bit signal. See S/N ratio.

amplitude, peak to peak • Measurement of the ordinate, i.e. (y) axis magnitude of a sine waveform depicted in the time domain from its greatest positive (+) value (crest) to its greatest negative (–) value (trough).

amplitude, RMS • Root mean square (RMS) amplitude is defined as the square root (√) of the mean, or average of the sum of all squared (V^2) values, where "V" represents all individual levels of a given signal. Adding the numbers in any group is the first step toward finding their average value. However, simply adding all levels of, e.g. a sine wave would equal (0) zero, because a sine wave is symmetrical about both x and y axes. Corresponding negative (–) and positive (+) values in a sine wave negate, or "cancel" each other in terms of averaging, because the sum of all these numbers is zero (0). However, squaring a negative number, i.e. multiplying a number by itself, produces a positive number. See relationship, square. Squaring makes it possible to add negative and positive numbers (such as sine wave signal levels) that have the same absolute value, avoiding their summing to zero (0). See absolute value. That is, without squaring, negative and positive levels of the same size would "cancel" to zero (0) when summed, and summing is the first operation of averaging. Squaring all levels makes it possible to account for negative levels while retaining a meaningful positive (+) sign. Averaging such squared numbers will therefore produce a positive result. Then, taking the square root of the sum of all squared numbers is the necessary inverse operation of squaring those numbers. These three steps effectively yield a positive number that represents the RMS value of all levels in the signal. • Root mean square (RMS) amplitude is easier to understand when you realize that actual math operations are taken in reverse of the left-to-right appearance of the letters R-M-S: (1) square (S) the value of every level, negative and positive; (2) find the average, or mean (M) by summing all these squared values, then dividing by the total number of squared values; (3) take the square root (R) of this average (mean) value to "undo" (take the inverse of) the squaring process (step 1). For example, if "V" represents the value of various levels named 1, 2, 3 . . . N, as differentiated using subscripts (graphically lowered numbers or letters), then: V RMS = $((V_1^2 + V_2^2 + V_3^2 \ldots + V_N^2) / N))^{1/2}$. [Note: taking the square root of a number, e.g.

"N" may be shown using a radical sign (√ N), or it may be depicted as: $(N)^{½}$]. Root mean square (RMS) is the ordinate-based (y) value used to measure or calculate power of a signal.

analog • An analog signal represents some physical phenomenon using continuous levels rather than discrete or numerically discontinuous means of representation. • A system that generates, models, simulates, processes, or transduces signals continuously, without use of quantized, or intentionally discrete values. Cf. digital.

analog module • Generator or processor of analog, i.e. continuous signal(s). • The eponymous analog, subtractive modular synthesizer (900 Series) designed in 1964 by the celebrated American physicist Dr. Robert A. Moog (1934–2005) exemplifies design of analog electronic sound synthesis modules. Thank you, Bob.

analog signal • A continuous signal, i.e. one that is not encoded in steps, reduced resolution, or discrete levels. Amplitude changes of an analog signal are represented continuously, rather than as discrete, encoded steps as in pulse code modulation (PCM), aka sampling. An analog signal has levels that are analogous to the phenomenon it represents and features electrical correspondences with the levels of that phenomenon. Such correspondences are made with no discontinuities between successive levels over time. For example, a sound pressure wave and its equivalent electrical signal at the output of a microphone (mic) are both continuous, and each is analogous to, or an analog of the other. As sound pressure level (SPL) fluctuates continuously, the mic's electrical output signal level mirrors these changes directly, proportionately, and continuously. • In contradistinction, a digital signal represented using pulse code modulation (PCM) exhibits no changes of actual signal levels that directly correspond to level changes of the original analog signal it represents. The varying levels of an analog signal are not represented by corresponding, or analogous level changes in a PCM digitized signal, but are encoded to represent numbers. A PCM (sampled) signal has only two states or levels, "high" or "low," and it is therefore a binary encoded signal. This is in contradistinction to the infinitely many different levels the original analog signal may exhibit.

anechoic • Literally, without echo. Conditions that produce absolutely no echo(es) of sound caused by reflections, are found in outer space. However, near-anechoic conditions are found in an appropriately constructed, highly absorptive (non-reflective) anechoic chamber used to test audio equipment and facilitate experiments with sound. Cf. echo.

a posteriori • (Latin) Means or intellectual basis of gaining understanding or knowledge from observable facts, direct experience, or reasoning from such facts and actual outcomes back to their root causes. Cf. a priori.

a priori • (Latin) Existing in thought prior to, or independent from real world knowledge gained due to systematic observation, empirical (objective) experience, or scientific experimentation. This expression implies "before," or prior to, pointing to having a conception or formulation of some idea before investigation of the actualities in a specific field. For example, the authors believe that any a priori classification of modules or signal functions prior to connecting module output(s) to actual audio, control, and/or timing (ACT) input(s) in a sound synthesis system, is an example of convergent thinking. For an example of an idea or schema based on a posteriori, or empirical experiment and experience rather than a priori surmises, see the glossary entry ACT. Cf. a posteriori.

ASCII • Acronym for American Standard Code for Information Interchange (ASCII, pronounced ask´ee), a communications standard that features binary encoding of letters of the English alphabet, numerals, and other symbols or characters. The computer peripheral on which one types to enter data is commonly called an alphanumeric, or ASCII keyboard, in contradistinction to, e.g. the keyboard of a piano. Each ASCII keyboard symbol or letter is coded using 7 bits of an eight (8) bit data word, or byte that can represent the basic set of symbols, featuring 94 printable characters, exclusive of the space. The remaining 33 ASCII characters are used for control functions, some of which are now outmoded. Any number of proprietary so-called “extensions” of ASCII use eight (8) or more bits to encode a larger number of characters, although ASCII per se legitimately remains a seven (7) bit standard. See alphanumeric. See byte.

arbitrary • Based on personal perception or feelings, rather than on known principles, accepted procedures, empirical evidence, or established technical protocols and standards. In particular, arbitrary decisions made by some software and hardware designers seem to ignore known principles of cognitive psychology, effective graphic design, ergonomics (human engineering), or even technical standards in the field. For example, knobs and other controls often provide arbitrary calibrations (e.g. 0–99 or 1–100), instead of globally accepted non-arbitrary calibrations such as frequency expressed in hertz (Hz), time in seconds (s) and milliseconds (msec or ms), and signal power in decibels (dB) or percentage of full scale (FS). See amplitude, full scale. See calibration. • Web page and other graphic designs that torture the human eye also manifest arbitrary decision making, e.g. placing miniscule magenta (red) letters on a black background. Puh-leeeeze! Learn something about graphic design prior to becoming a “web page designer.”

array • A group of items, quantities, symbols, or data arranged in a particular way, e.g. tabular form, a single line or column, or in a range of contiguous (adjacent, or consecutive) memory addresses in a computer. Cf. matrix.

attack • The initial part of an articulated (not droning) sound. See transients, attack. • The first segment, aka attack (A) produced by an ADSR envelope generator (EG). See ADSR. The attack segment of a typical ADSR envelope generator rises from zero (0) level to full scale (FS), i.e. the maximum allowable level, in an attack (A) time programmed by the user. In some designs, envelope generator attack (A) time may be subdivided into several (shorter) segments, each having two programmed level(s) and a single programmed time, facilitating shaping of the curve of the longer, overall attack (A) segment. See EG segment.

attenuation • Reduction or lessening of signal amplitude or level. Attenuation is a signal processing technique that reduces any signal level or amplitude toward zero (0), regardless of the processed signal's (±) polarity. • In digital signal processing (DSP) terms, numbers of either polarity (±) that represent signal level(s) are reduced toward zero proportionately, based on the percentage of attenuation, absolutely without regard to (±) sign. Attenuation reduces the absolute values, i.e. the magnitudes of the numbers that represent signal levels. See absolute value. Cf. amplification.

attenuator • Broadband signal processor that provides a fixed, but typically alterable reduction of amplitude or level of the signal passing through it. An ideal attenuator has a flat frequency response that does not "color," or change the processed signal's tonal balance. Theoretically speaking, such an attenuator reduces amplitudes of partials in all frequency bands proportionately. For example, a 1:2 negative gain, or "loss" is produced by a 50% attenuation of all partials processed by an ideal attenuator. • The concept of an "ideal" device is theoretical and is countered by the capabilities of actual designs and their resulting real world performance. See tonal balance. See frequency response, flat. Cf. amplifier.

audible frequency span • The frequency band bounded by the purported limit(s) of human hearing, 20 Hz – 20 kHz. See (measurement unit) hertz.

Aunty Em • The aunt of Miss Dorothy Gale of Kansas, USA, late of the Land of Oz. • See also (modulation) etc. such as FM, AM, PCM, PWM, PPM, etc. • It seems that we have Ems and Ms. And Mars (the candy company–not the planet) has M&Ms. These plurals may also be seen as Em's and M's and M&M's. Both forms are accepted, e.g. VCAs, or VCA's for more than one (1) VCA. The authors prefer the "VCAs" form, as neither of us is obsessively "possessive," particularly in the matter of collecting and subsequently distributing apostrophes (').

aural • Relating to hearing or the ears, particularly human perception of sound and music, or responsiveness to speech or that which is audible. Musicians should develop powerful aural (listening) and analysis skills. Cf. oral.

axis • Graphical dimension or direction depicted as a line, on or about which numbers or relative magnitudes are displayed. On typical two dimension graphs in audio engineering, e.g. frequency response curves, horizontal (x) axis (abscissa values), and vertical (y) axis (ordinate values) may converge toward zero (0) respectively at the lower left of the graph. The two dimensional graph known as (four quadrant I-IV) Cartesian space depicts zero for both axes in the center, aka as the origin. • The plural of axis is axes (pronounced aks´eez). Three dimensional space has three axes (x,y,z). The authors generally avoid space(s) that exceed three dimensions, notwithstanding the splendid work of Mr. Albert Einstein and Mr. M. C. Escher.

BAM • Balanced amplitude modulation (BAM), aka "ring" modulation. A type of amplitude modulation (AM) using a four quadrant multiplier, in which signals(s) at both carrier (C) and modulation (M) inputs are suppressed, i.e. do not appear at the multiplier's output. For a description of BAM output, see (modulation) AM. • Ring modulation is so-named due to early hardware designs that feature a near-circle, or diamond-shaped "ring" of four connected diodes. A diode is an electronic component that restricts electrical current flow to a single direction, i.e. one polarity (+) or (–).

balanced line or cable • See cable, balanced. Cf. cable, unbalanced.

balanced modulator • A four quadrant multiplier used for balanced amplitude modulation (BAM), aka "ring" modulation. See multiplier, four quadrant. Cf. multiplier, two quadrant. See BAM. See (modulation) AM.

band • A span of contiguous, or adjacent frequencies with a bandwidth, or frequency span defined by the highest and lowest frequencies at the extremities of the band. The term is often associated with filters, processors that attenuate or boost partials within selected frequency band(s). • Band: a loose association of musicians that typically disbands, particularly–and ironically, immediately after such a group first becomes commercially viable ("artistic" differences.)

band, pass • A pass band comprises a group of contiguous, or adjacent frequencies that pass through a processor, particularly a filter. • In a cutoff filter, one or more half-power point(s), aka cutoff frequenc(ies) define boundar(ies) between the pass band(s) and stop band(s). Cf. band, stop. See filter, etc.

band, stop • A stop band comprises a group of contiguous, or adjacent frequencies that have been stopped, or "cut off," i.e. attenuated by a processor, particularly a filter. One or more half-power point(s), aka cutoff frequenc(ies) define boundar(ies) between pass band(s) and stop band(s). Cf. band, pass. See filter, etc.

bandwidth • A span, or band of contiguous, i.e. adjacent frequencies bounded by the highest and lowest frequencies, often expressed as the difference in hertz (Hz) between those frequency extremes. For instance, the bandwidth of human hearing is approximately 19 980 Hz when that bandwidth is stipulated to be 20 Hz–20 kHz. • Bandwidth represents the limits of the frequencies that a module, communications device, or information system can generate, transmit, receive, or process. This is also equated to the speed at which data can be transmitted. (It takes more time to transmit a broader (higher) bandwidth, given a constant data rate).

(bandwidth) broadband • Broadband describes any device or module capable of generating (e.g. noise generator) or processing (e.g. amplifier, attenuator) the widest span of frequencies of interest (e.g. audio frequencies). The ideal frequency response of a broadband device is designed to be "flat." See frequency response, flat. • Capable of high speed data transfer. Cf. (bandwidth) narrowband.

(bandwidth) narrowband • Narrowband describes a device or module capable of generating (e.g. envelope generator) or processing (e.g. filter) a reduced span of frequencies of interest (e.g. audio frequencies). • Not capable of high speed data transfer, i.e. having a relatively low data transmission rate. Cf. (bandwidth) broadband.

base • The number of symbols in a positional ("base and place") counting system such as decimal (radix, or base of ten symbols 0–9). The base of a number system is aka as its radix. • See number base.

battery • A collection of cells that are connected in some configuration (series or parallel) to provide DC power. See cell.

baud, or "baud rate" • Signaling rate of a serial communications line or link, specifically the total count of "signal events" per second, e.g. the number of signal polarity changes from positive (+) to negative (–). The number of times per second a line can change its polarity. • Baud and bit rate are not synonymous, as more than one bit may be encoded within a signal event using phase shift, amplitude variation, or various other encoding techniques. For example, a modem transfers signals using a telephone line, which is limited (for an ordinary telephone line) to a baud of 2400 by the telephone company. Higher data throughputs achieved by modems advertised as higher than 2400 baud are accomplished not due to a higher baud, but by using data compression, frequency, and/or phase modulation techniques. • The Musical Instrument Digital Interface (MIDI) serial communications protocol (rev 1.0) debuted with a bit rate of 31 250 bits per second. Only when two serial (e.g. MIDI) devices are connected using direct cables are baud and bit rate (bits per second) indeed synonymous. To the extent that MIDI devices conform to such a connection restriction, then MIDI "baud rate" and bit rate are the same. Like many serial communications protocols, MIDI entails a "no parity (start bit), 8 data bits, "stop bit" format, yielding a total of 1 + 8 + 1 = 10 bits per "character." Whatever time period is allocated to send a character, the bit rate of such a serial protocol might be as much as 10 times the "baud rate." The term "baud rate" is actually redundant, and therefore poor usage, as baud inherently entails the concept of rate. Cf. bit rate. • The concept of baud was developed by Jean Maurice Emile Baudot, of the French Telegraphic Service, who devised a uniform length 5-bit code for characters of the alphabet during the late 19th century.

beats • Smooth, repetitive loudness pulsations created when two or more audible pitched tones with very similar frequencies are sounded simultaneously. The speed of beats is called the primary beat frequency, which is determined by the difference between the fundamental frequencies of the two tones. For example, if one instrument sounds 440 Hz, and another sounds 443 Hz (or 437 Hz), then 3 beats per second occur, i.e. a 3 Hz beat rate results. This is heard as three (3) loudness pulsations per second. When the individual frequencies of two tones progressively diverge, or grow farther apart, the speed of beats increases until individual beats can no longer be discerned. Such rapid beating may then be described in relative terms of "roughness," which relates to whether the interval(s) (distances between notes) produced are judged to be consonant ("pleasing to hear, or concordant") or dissonant ("unstable, unpleasant, or in need of resolution to a consonant interval.") • Beats are produced due to alternating conditions of constructive (louder) and destructive (quieter) interference of waves, i.e. when ongoing instantaneous levels of at least two sound waves with very close fundamental frequencies sum algebraically (±) over time. Beats occur when a multiplicity of tone generators are purposely detuned slightly (e.g. multiple strings of the middle notes on the piano, or dual strings on a 12-string guitar). When the beats of many such slightly detuned "voices" occur, this creates chorus effect, as heard in an a cappella choir (singing group that performs without instrumental accompaniment), violin sections in symphony orchestras, etc. Chorus effect(s) may also be created using electronic processors. See chorus effect. Cf. tremolo.

beat, zero • A wave or signal interference (algebraic (±) summation) that produces no beats. See beats. • Two periodic signals or pitched sounds tuned to precisely the same frequency, or to a "just" temperament musical interval (i.e. whole number frequency ratios such as 3:2, e.g. the perfect fifths (P5) of properly tuned violin strings) produce zero beats, aka a zero beat condition.

Bel • A unit of measurement for objective comparison of two powers. A power ratio expressed logarithmically using "common," or base ten powers. Cf. decibel, the more commonly used unit. • The formula for sound intensity level (SIL), or power in Bels is: $SIL_{Bel} = \log (W_1 / W_0)$ where W_1 and W_0 are powers measured in watts (W). See watt. Letters I_1 / I_0 for intensities, or P_1 / P_0 for powers are also seen in this formula in lieu of the "W_1 / W_0" designation. Subscripts ($_1$) and ($_0$) here act only as names that differentiate, such as a family name followed by "Sr." or "Jr." • The Bel is subdivided into 10 decibels (dB). See (measurement unit) decibel.

beta test • Tryout or "test" of hardware or software by a limited number of valued or respected customers or professional in the field, prior to general release of a product to the public. This is a "field" test by people who are not employees of the company involved. Many software "bugs" are discovered by beta testers, who may exercise product features in ways unintended or not thought of by the product designers.

bias, DC • A step signal, i.e. a fixed, but typically programmable level of direct current (DC), particularly in an analog system. A bias is typically used as a control signal to set initial, or nominal operating condition(s) of a parameter on a module, e.g. filter cutoff frequency. For instance, audio oscillator frequency is typically internally biased to sound 440 Hz when note A4 (the A above "middle C") on an associated keyboard controller is played. A bias is a signal processor, i.e. an attenuator whose DC input signal is typically derived from the system power supply. • In virtual terms, such a DC-like signal may be known as a constant. In terms of modules, a constant is a signal generator, requiring no signal "source" such as DC, that is subsequently processed (attenuated). Because a constant module is a generator, it has no signal input, only a signal output that directly generates a number. When a constant, or virtual representation of a step signal (bias) is involved, the authors recommend use of the verb form offset, in lieu of older analog system terms "bias" or "biased." See offset. • The earliest form of analog magnetic audio tape recorder bias used direct current (DC), but DC bias was superseded by AC bias. See bias, AC.

bias, AC • A fixed frequency sine wave (alternating current, or AC) in the range of 100 000 Hz that is mixed, i.e. added to the recorded signal, in order to improve the otherwise nonlinear response of the head(s) of a tape recorder. In particular, analog audio tape recorder head(s) have a severe nonlinear response (input versus output characteristic), which would, without addition of an AC bias signal, distort recorded signals badly. The addition of an alternating current (AC) bias moves the program, or signal to be recorded into linear regions of the frequency response curve of the head. This high frequency AC bias signal is "trapped" by a filter and removed upon playback. See distortion, nonlinear. Cf. linear. • The earliest form of analog magnetic audio tape recorder bias used direct current (DC), but DC bias was superseded by AC bias. Cf. bias, DC.

binary • Having two states or conditions, often represented numerically using zero (0) or one (1). • A number system with a radix of two, using symbols zero (0) and one (1) for counting and/or performing arithmetic. See number base, etc.

binaural • A audio recording and playback system with left and right (L-R) channel recordings made using a dummy "human head" fitted with microphones where (L-R) ears would be located. A binaural system provides particularly realistic sound location cues when such recordings are monitored using stereo headphones. The putative binaural localization advantage is reduced if stereophonic (L-R) loudspeakers are used for playback rather than stereophonic headphones. Cf. stereophonic.

bipolar • See (polarity) bipolar. Cf. (polarity) unipolar.

bit • Contraction of "binary digit," a bit is a character used to represent one of the two digits that comprise information in a digital computer, binary encoded information system, or physical entity or system element having only two possible states. During a given moment, a bit has only one of two possible states, conceptually used to represent the numerals 0 or 1. • A bit is not a virtual entity within a digital computer. Bit status ("0" or "1") is represented by an electrical signal on a wire or circuit board trace, a location on a hard disk with a specific magnetic polarization, a physical optoelectronic status, or by similar alternatives of the status of signals and/or system components, particularly those in a digital memory. Therefore, there are actually no "zero" (0) or "one" (1) numerals within a digital computer, common analogies and metaphors to the contrary! In the real world in which a computer resides, a bit is physical–not virtual. However, in the world of math a bit is quintessentially virtual–it represents a choice between the numerals 0 or 1. To discover the many protocols and forms of binary (two state) signals used to represent information, is to appreciate the distinction between physical signal characteristics, and their associated virtual binary representations. A "bit" is conceptually a single idea in mathematics. But the types of signals and protocols used to convey information using binary encoding are many and varied (e.g. pulse code modulation (PCM), biphase modulation, return to zero (RTZ), non-return to zero, etc.) Binary representations are often technically more involved than the popular but naive notion of "high = 1" and "low = 0" as the sole form that a binary signal can take. See virtual reality.

bit rate (serial) • The number of bits that can be sent via a serial communications line or link. The signaling rate of a serial communications line or link, specifically the total count of "signal events" per second, e.g. the number of signal polarity changes from positive (+) to negative (–), or baud does not constitute "bit rate." The number of times per second a line can change its polarity is the "baud rate." See baud. • Bit rate and baud are not synonymous, as more than one bit may be encoded within a signal event using phase shift, amplitude variation, or various other encoding techniques. For example, a modem transfers signals using a telephone line, which is limited (for an ordinary old fashioned telephone line) to a baud of 2400 by the telephone company. Higher data throughputs achieved by modems advertised as higher than 2400 baud are accomplished not due to a higher baud, but by using data compression, frequency, and/or phase modulation techniques. See modem. • The Musical Instrument Digital Interface (MIDI) serial communications protocol (rev 1.0) debuted with a bit rate of 31 250 bits per second. Only when two serial (e.g. MIDI) devices are connected using direct cables are baud and bit rate (bits per second) indeed synonymous. To the extent that MIDI devices conform to such a connection restriction, then MIDI "baud rate" and bit rate are the same. Like many serial communications protocols, MIDI entails a "no parity (start bit), 8 data bits, stop bit" format, yielding a total of 1 + 8 + 1 = 10 bits per "character." Whatever time period is allocated to send a character, the bit rate of such a serial protocol might be as much as 10 times the "baud rate." The expression "baud rate" is (harmlessly) redundant, and therefore poor usage, as baud inherently embodies the concept of rate. Cf. baud.

block diagram • A grouping of icons that represent modules connected, or "patched" into a particular configuration. In this context, an icon is a line drawing that represents a particular module. A module is a recognizable hardware or virtual "building block;" a unit (oscillator, filter, amplifier, envelope generator, keyboard, etc.) that can be connected to others to form various structures or subsystems within a modular system. The block diagram itself is often referred to as a patch, even though a block diagram is only a graphic representation of the connections among physical or virtual modules that constitute a particular patch.

Boolean operations • Decisions made, or computer code branching effected using logic gates, where each logic operand (input) and result (output) can have only one of two possible (binary) values at any specific time, 0 or 1. See logic gate. Named after George Boole (1815–1864), English mathematician and logician, who developed symbolic logic.

BPM • Acronym for beats per minute (BPM), expressed as a number, similar to the older metronomic mark (MM), which represents beats per minute. Spoken by saying the letters "B-P-M" serially. See clock. See metronome.

brick wall filter • Low pass filter with an extremely steep slope, i.e. –90 dB/Octave or greater. Such filters precede the analog to digital converter (ADC), and follow the digital to analog converter (DAC) in a sampling system. See filter, anti-aliasing. See filter, smoothing.

buffer • Computer data processing routine, and memory sectors reserved or actually used for allocating, scheduling, interchanging, or temporarily storing data, in order to compensate for different rates of data flow among system elements in a digital computer. A temporary storage space for data. • Analog device, or "isolating" analog circuitry that electrically protects a circuit from being adversely influenced by another circuit connected to it, due to a variety of electrical conditions such as impedance. Inputs and outputs with such circuitry are "buffered."

bus or buss • Circuit traces or wires designed to carry signals, particularly summation(s) of multiple signals, e.g. output bus, monitor bus, etc. on an audio recording console. • A channel or path comprising many discrete circuit board traces or wires that are neither physically nor electrically interconnected, designed particularly for parallel data transfer in a digital computer between major subsections, e.g. central processing unit (CPU), memory, peripheral (input/output) device(s), etc.

byte • A group of contiguous (adjacent) bits constituting a data word, or structure for representation of data larger than a bit in a digital computer. For example, eight (8) bits may constitute a byte used to represent a numeral or letter. See ASCII. Cf. bit.

cable, balanced • A balanced cable is a flexible electrical three (3) conductor system with two (2) wires that convey current carrying a signal of interest, e.g. audio. These two (2) wires are insulated from each other, and are typically enclosed within the third conductor, a metallic "shielding" mesh. This shield does not carry the audio signal, because it is not electrically connected to the two (2) enclosed wires that do. However, the shield does protect the cable from stray electrical interference, e.g. electrostatic fields that might otherwise induce interference in the signal-carrying conductors. In a balanced cable this shield may be connected to electrical ground (see immediately below), or not. When the shield of a balanced cable is not connected to ground, the cable and devices connected to it are referred to as being "isolated" or "floating." The two (2) wires and metallic braided mesh of a balanced cable are enclosed by a non-conductive fabric and/or rubber exterior covering. A cable is referred to as balanced when its design "balances" induced electromagnetic interference, e.g. hum at 60 Hz (USA), due to placing such an induced signal at opposing electrical polarities (±) within the two (2) wires that carry the signal. The two (2) wires that carry the signal in a balanced cable are typically placed as close to each other as possible in a twisted pair configuration. Each half-twist in a twisted pair causes the unwanted interference (hum) to appear at opposite (±) polarities on the two (2) wires carrying the audio signal. Due to the resulting destructive interference (algebraic summation), hum is substantially canceled by such a twisted pair configuration of wires. Cf. cable, unbalanced. See interference, destructive. • Any electrical signal requires two (2) conductive paths for transmission. For example, an incandescent light bulb has two wires or paths, one for input of current, and one that allows current to leave, a so-called "return" wire. Electrical power cables have two (insulated) wires that require no electrical shielding, as no intelligence, or signal other that power is intended to be transmitted. It was discovered in early telegraphy that the earth itself could function as one of the two (2) requisite "wires," and this "return" configuration was known literally as "ground" or "earth." But, use of the earth to ground devices is very inefficient, due to attendant substantial power losses. Nevertheless, telegraphy did work well, and required only one (1) wire to transmit intelligent signals, due to use of the earth itself as the other (ground) “wire” (return path). Earth grounding is not always practicable for power transmission, audio engineering, or modern electronic communications, due to the earth’s high resistance to the flow of electricity, which would cause large signal losses. Unlike telegraphy’s single transmission wire, power transmitted to homes arrives on two wires that are not grounded to earth. And, in various audio circuits and their interconnecting cables, grounding is typically not to earth. Instead, one of the two (2) conductors required for current flow is “grounded” by connecting it to the chassis (rhymes with “classy”) of the device or circuit involved. The chassis may comprise the conductive substrate of the circuit board(s), or a conductive supporting structure that encloses electronic components, typically a metal box that houses the entire electronic device. This type of grounding is a “single-ended,” or unbalanced connection, because ground is connected to one of the two (2) “wires” needed to carry the signal, e.g. audio. Unbalanced cable(s) use this type of grounding. See cable, unbalanced. • An unbalanced cable works well over short distances where the voltage of the signal being transmitted is large with respect to any induced interference such as hum. For example, the cables used to connect modules on the vintage Moog analog voltage controlled synthesizer are unbalanced. But, especially over longer distances, smaller signals such as those produced by microphones or electric guitars, are significantly degraded by interference induced in such unbalanced cable(s). Therefore, a balanced cable is used, where grounding is not involved with carrying the intelligent, i.e. audio signal. • The understanding of balanced vs. unbalanced cables is confounded by false correspondences, particularly the commonly heard misnomer “high impedance” vs. “low impedance” cables. Impedance (Z) is the sum of several types of resistance to the flow of electricity, and no cable per se that carries audio signals is designed to resist signal flow in the least. The utility of such “high” impedance would be (?). To provide perspective, extremely high impedance “cables” that strongly resist the flow of electricity are exemplified by the heating element(s) in an electric stove! The plugs on respective balanced vs. unbalanced audio cables have become associated over time with high vs. low impedance circuits or devices to which such cables are typically connected. This has given rise to the misnomer “high impedance” versus “low impedance” cable. It’s the connectors (e.g. XLR for balanced “high impedance” cables, and ¼ inch phone for unbalanced “low impedance” cables) that have been associated with devices of different impedance–not the cables as such. This misnomer is sufficiently widespread that it’s unlikely to change–like the idea that all people with red hair have a bad temper. (If you lived all alone in a world of idiots, you would be cranky too!) See cable, unbalanced.

cable, high impedance • A misnomer for a balanced cable. See cable, balanced. • See impedance.

cable, low impedance • A misnomer for an unbalanced cable. See cable, unbalanced. • See impedance.

cable, unbalanced • An unbalanced cable is a flexible two (2) conductor electrical system comprising one (1) wire and one (1) metallic shield (or a “ground” wire). An unbalanced cable may be used to convey electrical current that carries a signal of interest, e.g. audio. The single (1) wire is typically enclosed with, but insulated from the other conductor, the ground wire. The mechanical construction of the shield is typically a metallic mesh. This ground and wire together carry the audio signal because they are both electrically connected to circuit elements. The shield of an unbalanced cable may be connected to electrical ground. (Grounding is discussed below). The two conductors of an unbalanced cable are enclosed by a non-conductive fabric and/or rubber sheathing. Interference such as 60 Hz (USA) hum may inadvertently be induced in an unbalanced cable due to its proximity to nearby alternating current (AC) power cords or devices powered by AC. The unbalanced cable lacks the twisted pair signal conductors of the balanced cable, that cancel hum. See cable, balanced. • Any electrical signal requires two (2) conductive paths for transmission. For example, an incandescent light bulb has two (2) wires, one (1) for input of current, and a second wire (2) that lets current leave–a so-called "return" wire. Electrical power cables have two insulated wires that require no shielding, as no signal other that power is transmitted. It was discovered in early telegraphy that the earth itself could function as one of the two (2) requisite "wires," and this "return" configuration was known literally as "ground," or "earth." However, use of the earth to ground devices is inefficient due to large power losses. Nevertheless, telegraphy did work, and required only one (1) wire to transmit intelligent signals, due to use of the earth itself as the other "wire" (return path). Earth grounding is not practicable for power transmission, audio engineering, or electronic communications, due to the earth's high resistance to the flow of electricity, which would cause large signal losses. Therefore, unlike telegraphy's single transmission wire, power transmitted to homes arrives on two (insulated) wires, neither of which is grounded to earth. Also, in various audio circuits and their interconnecting cables, grounding is not to earth. Instead, one of the two (2) conductors required for electrical flow is "grounded" by connecting it to the chassis (rhymes with "classy") of the device or circuit involved. The chassis is the conductive substrate of the circuit board(s), or the conductive structure that supports electronic components, or a metal box that houses the entire electronic device. This is a "single-ended," or unbalanced connection, because ground is one of the two (2) "wires" needed to carry the intelligent signal, e.g. audio. • Unbalanced cable(s) work well over short distances where the voltage of the intelligent signal is large with respect to any induced interference such as hum. For example, the cables used to connect modules on the early Moog analog voltage controlled synthesizer are unbalanced. But, especially over longer distances, smaller signals such as those produced by microphones, are degraded by interference induced in such unbalanced cable(s). Therefore, a balanced cable is used. See cable, balanced. • The understanding of balanced vs. unbalanced cables is confounded by false correspondences, particularly the misnomer "high impedance" vs. "low impedance" cables. Impedance (Z) is the sum of several forms of resistance to the flow of electricity, and no cable per se that carries audio signals is designed to resist signal flow in the least. The utility of such impedance (resistance to the flow of the audio signal) would be (?). To provide perspective, extremely high impedance "cables" that strongly resist the flow of electricity are exemplified by the heating element(s) in an electric stove! The plugs on respective balanced vs. unbalanced audio cables have become associated over time with high vs. low impedance circuits or devices to which such cables are typically connected. This has given rise to the misnomer "high impedance" versus "low impedance" cable. It's the connectors that have been associated with devices of different impedance–not the cables themselves. This misnomer is sufficiently widespread, so it's unlikely to change–like the idea that all people with red hair have a bad temper. (If you lived all alone in a world of idiots, you would be cranky too!) Cf. cable, balanced.

cache • See memory, cache.

calibration(s) • Procedures, tests, or measurements that reveal deviations from an accurate standard, executed in order to correct errors or bring the tested system into alignment with that standard. • Numerical and/or graphical markings that indicate relative unit(s) of measurement adjacent to or around a knob, slider, virtual display, or visual indicator on a module or signal measuring device. For example, the gain control (volume knob) on a guitar amp is typically calibrated arbitrarily from one to ten (1–10), rather than in internationally accepted measurements of power such as the decibel (dB). See arbitrary.

Cartesian coordinate • Most often associated with the location system for a two dimensional graph. One of two values, each of which may be negative or positive, that create a coordinate pair (x,y) that locates a specific point on a two dimensional geometric plane whose axes are oriented at a perpendicular (right, or 90° angle), with a common zero point (origin) for both axes. A Cartesian plane is one having all possible points defined by such Cartesian coordinates, representing two dimensional space. Three dimensional Cartesian space has three coordinates (x,y,z). Cartesian space, whether two or three dimensional, relates to this coordinate system, as developed by 17th century French mathematician and philosopher René Descartes.

cascade • To connect modules, devices, units of data, or circuits in series, i.e. consecutively one after another. For example, four (4) simple one pole (–6 dB/Octave) filters connected in series (cascaded), create a filter with a steeper four pole (–24 dB/Octave) slope. • (verb) To make a series connection. • (noun) A connection made in series.

cell • The familiar small individual unit that provides DC power due to its electrochemical action, a multiplicity of which are typically connected in series or parallel to form a battery. A cell is often erroneously referred to as a "battery," but a battery comprises several cells electrically connected into some configuration. See battery. Cells connected in series (+ to – to + to – and so forth) provide a battery with higher voltage; in parallel (all + connected to each other and all – connected to each other), a higher current (amperage).

cent • One of one hundred (100) subdivisions of the equitempered musical frequency ratio (1.05946:1) that represents a half step (semitone). One (1) cent is one hundredth (1/100) of an equitempered half step, where the frequency ratio of a half step is the twelfth root of two, i.e. $2^{1/12}$ which is a ratio of approximately 1.05946:1, which rounds to approximately a 6% change. Therefore, 1200 cents is an octave (i.e. 12 half steps). •

The cent provides a means of describing various small intervals that comprise scales or temperament systems (subdivisions of larger intervals) in music. Both the cent and musical interval are based on constant percentage frequency changes that can be expressed as ratios, so both numbers of cents and associated musical intervals can be added directly. For example, fifty (50) cents is ½ of a semitone, a "quarter tone." Four quarter tones @ 50 cents apiece create the interval of a major second (200 cents). For larger intervals, e.g. a perfect fifth (P5) is seven half steps, which is 700 cents. Two (successive) melodic perfect fifths (P5) would equal (700 + 700 = 1400) cents, spanning the musical interval of a major ninth (M9). Cf. temperament.

cf. • Abbreviation that means "compare with," after the Latin "confer." Its use in this glossary indicates that the reader should look at other specified entries for a useful comparison or contrast.

channel • Physical path or virtual means of isolating and/or differentiating signals or data streams from each other. • A multichannel analog tape recorder, or its computer hard disk virtual equivalent, provide separate channels, aka "tracks" that facilitate individual manipulation of signals or streams of representations of numbers. In general, the number of "channels" may be limited by overall system (hardware) features, and the number of "tracks" likely manifests bandwidth (software) limitations.

chorus effect • Sonic attribute typified by a multiplicity of voices (human, instrumental, or electronic) that feature small, dynamic detuning(s), or frequency deviations among those voices. See beats.

clangorous • From clangor, a "loud ringing or banging," typified by a metallic, clanking, or bell-like sound. A clangorous sound typically includes inharmonics in its spectrum, so-called discordant" partials not harmonic to other partials. Inharmonics in a spectrum tend to obscure perception of a clear pitch. The spectrum of a bell typically has such inharmonics, causing difficulties in judging the pitch of the bell. See spectrum, line. See (partial) inharmonic. Cf. (partial) harmonic.

clock • Device that subdivides the passage of real, or absolute time in hours, minutes, seconds, milliseconds, etc. • A module that precisely maintains the (programmable) frequency of its periodic output signal. An oscillator may be used as a clock. This type of clock might be used to arbitrarily mark or subdivide absolute time into relative periods, or units of relative musical time. Musical tempo is a relative subdivision of a unit of real, or absolute time, e.g. 120 beats per minute, or "march tempo" (i.e. metronomic marking (MM) = 120). Metronomic markings are also known as beats per minute (BPM). See metronome. See BPM.

clock, MIDI

clock, word

C:M ratio • Carrier to Modulator ("C to M") ratio, a ratiometric representation of the respective frequencies of a carrier (C) signal and its associated modulation (M) signal. In music synthesis, the C to M frequency ratio (C:M) determines the potential spectrum, i.e. the frequencies of the set of partials produced by full scale (deepest) amplitude modulation (AM) or full scale linear frequency modulation (FM), i.e. rapid (audio frequency) modulation techniques. See spectrum, potential. See full scale. See index of modulation. • An FM (frequency modulation) C:M ratio of 1:2 indicates that the carrier (C) frequency is one half (½) that of its associated modulation (M) frequency. Conversely, in this example, modulator (M) frequency is twice (2 times) that of the carrier (C). This 1:2 C:M ratio produces a potential spectrum of only odd harmonics (H1, H3, H5, H7, etc.) A C:M ratio of 1:1 produces a potential spectrum of all harmonics (H1, H2, H3, H4, etc.) • Integral C:M ratios, i.e. those for which both C and M are whole numbers (or resolve to an integral relationship), produce only harmonics when rapid amplitude modulation (AM) or linear frequency modulation (FM) techniques are effected. (The C:M ratio is integral if it is manifestly a ratio of whole numbers, or can be resolved to a rational number, i.e. a whole number ratio. For instance, a ratio of 0.5:1 is equivalent to 1:2, and is therefore behaves like an integral C:M ratio in linear FM. However, the ratio 0.5:1 is an octave lower than (1:2). A non-integral, or irrational C:M ratio (e.g. 1:3.14159), or "one to pi," creates a potential spectrum that includes inharmonics, aka "nonharmonics." See FM potential spectrum. See (partial) inharmonic. Cf. (partial) harmonic. • Unfortunately, the proper form of a linear FM ratio (C:M) is sometimes seen as (M:C) in literature in the field, which reverses the sense of the relationship between carrier and modulator. The developers of linear FM synthesis, John Chowning et al, specified C:M as the appropriate form for expressing the frequency ratio between FM operators ("oscillators"). Follow suit.

cocktail party effect • Characterization of the ability to focus listening on distant conversations or sounds of greater interest while in the presence of closer, more intense sounds. (The eyes glaze over when this skill is exercised at an actual (not virtual) cocktail party. Ironically, ingestion of alcohol does not enhance cocktail party effect, despite what the name may imply.) This perceptual skill, when properly developed, illustrates that an audio engineer can focus on low level sonic details in a complicated sound mix that inexperienced or untrained ears would not discern. Ability to focus on tiny noises in a sound or distortion in a mix is also a related example of this skill. Most people have this capability to some extent, but audio professionals develop this skill to the extreme.

code • Symbols or protocols using symbols that purposely cause communication to be cryptic; be devoid of any apparent surface meaning; or to convey an apparently innocuous meaning that actually masks a covert (hidden) message. • Digital computer software is aka code, probably because its language, syntax, and meaning are occult, i.e. hidden from the uninitiated.

coefficient • A number (e.g. 2) placed before a letter (e.g. x) that is a variable in algebra, to show multiplication of variable x by coefficient 2, or (2x). Taken together (2x), the coefficient and variable constitute a term. • Terms with coefficients for amplitude and frequency respectively, can be summed to construct a given geometric waveform. A sawtooth waveform has partials (sine waves) at all frequencies of any harmonic series, i.e. sinx, sin2x, sin3x, sin4x, sin5x, etc., where "x" is the fundamental frequency and the numbers are coefficients that represent specific harmonic numbers. These coefficients represent harmonics, or whole number (integer) multiples of the first harmonic (H1). The first harmonic (H1), aka the fundamental, is shown simply as "x" (the coefficient "1" for "1x" being understood). • There can be amplitude coefficients in such a series of terms as well. The amplitude of any harmonic in a sawtooth waveform, with respect to the amplitude of its first harmonic (H1), is the reciprocal (inverse) of the selected harmonic's number. For example, the amplitude of the 5th harmonic of a sawtooth waveform is 1/5th that of the fundamental (H1). Therefore, a sawtooth waveform "y" can be constituted by the formula: y = sinx + 1/2 sin2x + 1/3 sin3x + 1/4 sin4x + 1/5 sin5x + etc. Each fraction (1/2, 1/3, 1/4, 1/5, etc.) is a coefficient that represents the amplitude of an individual harmonic (sine wave). The fundamental frequency here (1/1sin1x, simplified to "sinx") is also the first harmonic (H1), notwithstanding casual use of the term "harmonic" by stringed instrument players to mean a note necessarily higher than the frequency of an open string. The term harmonic represents all of the integral, or counting number frequency relationships in a harmonic series, including the fundamental, which is correctly called the first harmonic (H1). Eschew obfuscation! Avoid the outmoded term "overtone" like the plague, or at least a serious pox. Use the modern term partial. See (partial) harmonic. See (partial) inharmonic. See synthesis, additive.

coloration • Alteration of the tonal balance of an audio signal, particularly due to unequal change(s) of signal characteristic(s) such as the amplitudes of partials that make up a complex waveform. Significant coloration causes a perceived change of timbre, or tone color. Cf. frequency response, flat. See complex waveform.

complex waveform • Representation in the time domain of a signal that comprises more than one sine wave, or partial. The algebraic summation of several partials constitutes a more complicated, i.e. complex waveform. Acoustic musical instruments produce complex waveforms. • In contrast, a sine (or cosine) wave generated by an oscillator is a "simple" waveform–a single partial. See waveform, etc.

comparator • Electronic circuit or its virtual equivalent that compares two instantaneous input signal levels (or numbers), and outputs a signal that indicates whether one of the signals is: greater than (>); less than (<); or equal to (=) the other. The comparator is used in analog electronic feedback circuits, and is also an important tool for conditional branching and decision making ("if, then: else" operations) in logic circuits, logic gates, or algorithms in digital systems such as the computer. See logic gate.

compression, audio • Techniques that reduce the dynamic range of audio signals in order to improve recorded or transmitted signal to noise (S/N) ratio, prevent distortion, and/or optimize transmission and/or storage of audio signals in an (e.g. audio) recording system. • Audio compression techniques generally either: (1) compress or limit high levels (e.g. DBX noise reduction); or (2) boost low levels (e.g. Dolby noise reduction) prior to recording, in order to compress (reduce) dynamic range. This reduced dynamic range is typically expanded (increased) upon playback of the recorded signal(s). The complementary process of compressing and expanding is aka companding. Companding an analog signal improves signal to noise (S/N) ratio by effectively lowering the "noise floor" of broadband noise such as tape hiss, e.g. in analog tape recordings. • Audio compression in a digital system limits dynamic range of a program signal in order to reduce quantization noise, which has a narrower bandwidth than broadband noise, but is nevertheless highly irritating to the nerves. Broadband noise (e.g. tape hiss) is inherently eliminated in digital sampling recording systems. But, quantization noise is created by sampling, particularly when a low number of bits is used to quantize signal levels. Use of more bits (e.g. 20 or 24 bit sampling) putatively reduces quantization noise to negligible levels. See noise, quantization. See compression, data.

compression, data • Reduction of the bandwidth, baud, or number of bits required to manipulate, transmit, or store information in a digital system. • Data compression and quantization noise in a digitized audio file have an inverse relationship; i.e. as digital file data are compressed (bit rate reduced), quantization error increases. • Data compression techniques are of two types: (1) lossy, which loses data due to compression; and (2) lossless, which does not lose data deemed to be discernible, "important," or "relevant" due to such data compression. • Many data compression techniques, e.g. compression of video for DVD or HDVD, depend on "delta" algorithms that store only the data that actually changes from moment to moment, and append this data to that which is redundant (non-changing) from one instant or time frame to the next.

(computer) application • Higher level or more global set of software commands or "code" designed to accomplish a particular task, e.g. word processing, digital audio recording, MIDI sequencing, etc.

(computer) CPU • Central processing unit (CPU), the circuitry in a computer designed to interpret and execute instructions from software or firmware, or commands from other circuitry within the computer.

(computer) firmware • Software that has been stored in a physical electronic memory component, or "chip."

(computer) hardware • Mechanical and electromechanical (physical) devices that constitute a computer, e.g. disk drive(s), monitor(s), central processing unit (CPU), electronic components, etc.

(computer) machine language • Lower level commands or instructions derived from higher level computer software. See (computer) software. Machine language comprises strings of representations of zeroes (0) and ones (1) that directly manipulate bits and bit structures within a digital computer.

(computer) peripheral • A device (e.g. disk drive, printer, video monitor, etc.) external to, but connected to and controlled by the computer's central processing unit (CPU).

(computer) software • Commands, or "code" written in a computer language (e.g. FORTRAN, BASIC, FORTH, LISP, etc.) that exercise computer hardware in order to accomplish a task. See code.

concatenate • To link system elements or information units serially into a single unit, sequence, or chain. • To connect modules or circuits in series. For instance, several one pole (–6dB/Octave) filters may be concatenated to produce a four pole (–24 dB/Octave) filter with a steeper slope. See connection, series. See cascade.

connection, parallel • Module, circuit, or component configuration with connections of a given signal simultaneously to all inputs in that group. A signal can pass through any member of the group independently, without consideration of the input/output (I/O) status of the other units in the group. Cf. connection, series.

connection, series • Module, circuit, or component configuration in which a signal must pass through one member of a group prior to being connected to the others. The signal cannot pass through any module or unit in the group independently, but is constrained to pass through units serially, i.e. one after another–in the order dictated by the designer or programmer. The signal can appear at the output of a given module or member in the chain only after it has passed through the inputs and outputs of all preceding members in that chain. Cf. connection, parallel.

constant • A quantity, or number that retains the same value during a specified set of mathematical calculations or scientific experiments. Cf. variable. • A constant is often a known property or condition that can be assumed to remain at the same value during calculations or experiments, e.g. the velocity of sound in air: 344 m/s @ 20 ° Centigrade, or Celsius. (Three hundred forty four meters per second at twenty degrees Centigrade). When conditions change, such a constant may also change, e.g. the velocity of sound in air @ 30 ° Centigrade is approximately 350 m/s. That is, there is a "delta," or change of velocity of 6 m/s per 10 ° Centigrade. (Warmer is faster). Therefore, the term "constant" does not necessarily mean "unchanging," or immutable in a mathematical context. • In terms of modules, a constant is a virtual generator with an output that functions similarly to a hardware processor called a bias. A bias is a hardware processor with a fixed full scale direct current (DC) signal permanently attached to its signal input; this DC signal passes through an associated attenuator to yield a specific bias level (step signal, typically a voltage level) at the signal output. In contradistinction, a constant is a virtual generator capable of directly producing a step signal whose size or level is represented numerically. Cf. bias, DC. See attenuator.

contiguous • A description of elements that are situated next to, adjacent to, or arrayed in close proximity to each other. A frequency band is a group of frequencies that are contiguous to each other. e.g. the frequencies in the frequency band from 1 kHz to 2 kHz.

convergent • Regarding thinking, a tendency to conform to well-established patterns, tropes, truisms, homilies, and conventions. Convergent thinking is so prevalent, that “non-convergent” doesn’t appear in most dictionaries! The antonym “divergent” must suffice.

(converter) ADC • Analog to digital converter (ADC), an electronic circuit or mechanism that changes a continuous (analog) signal into an equivalent discrete (digital) signal. The ADC serially samples, or captures the instantaneous level of a continuous signal in time, and provides a string of numerical representations (samples) of those levels that may be stored in computer memory. See filter, anti-aliasing.

(converter) DAC • Digital to analog converter (DAC, pronounced dak), an electronic circuit or mechanism that changes a discrete (digital) signal into an equivalent continuous (analog) signal. The DAC processes serially retrieved discrete (digital) values from computer memory or a recording medium (compact disc (CD), DVD, etc.), and changes this string of numerical representations (aka samples) into a continuous (analog) signal, i.e. the recovered, or reconstituted audio signal in a sampling system. See filter, smoothing.

convolution • Convolution describes, for any linear system (amps, filters, etc.), the interaction between the input signal and the impulse response of the system. That is, signal output is a function of the input signal and the characteristics of the processing device. Implicit uses yield effects such as filtering, reverberation, etc. • Explicit digital signal processing (DSP) techniques of convolution, aka “cross-synthesis,” involve two digital audio files “a” and “b,” where each sample of “a” is multiplied by every sample of “b.” Signal output is the sum (mathematical infinite integral) of all the resulting arrays produced by such multiplications, each having a length “b” for every sample of “a.” This brute force technique is called direct convolution. Allied techniques such as fast or sectioned convolution are less computationally intensive, making implementation practicable. Convolution of two audio signals is effectively filtering one sound’s spectrum by the other sound’s spectrum, which may create the sound of one instrument “playing” the other. In general, only those frequencies prominent in both “a” and “b” files can appear prominently in the convolved output. • Multiplication of two audio frequency signals in the time domain (e.g. amplitude (AM) or ring (BAM) modulation), is equivalent to convolving their spectra in the frequency domain. But, in the case of simple multiplication (AM) of two (time domain) waveforms, the math has less formidable computational overhead. Considering the digital sound files for such (AM, BAM) modulation techniques, or time domain multiplication, each sample “a” is multiplied only by the single corresponding sample “b,” not each sample “a” multiplied by every sample “b” as in direct convolution.

coordinate pair • Two values (x,y) that represent a single point, i.e. one discrete location on a Cartesian plane, a typical two dimensional graph on a single plane. The (x) values lie on the horizontal dimension. The (y) values lie on the vertical dimension. A coordinate pair of such values is shown in alphabetical order, separated by a comma (x,y), that is: (horizontal, vertical). The origin or zero (0) value points for each axis typically intersect in the center of the graph having all four quadrants I, II, III, IV. Positive (x) values are located to the right of the origin, negative to the left. Positive (y) values are located above the origin, negative below. See axis. • A graph depicting the relationship of two variables, i.e. a line, curve, or transfer function typically depicts many individual points, or a theoretically infinite number of coordinate pairs. See abscissa. See ordinate.

correlate • (noun, core´uh let) One of at least two things, ideas, variables, or properties that exhibit complementary correspondences, i.e. mutual relationships. The objectively measured audio signal characteristic intensity, and its subjectively judged sonic attribute loudness, are correlates to some extent. However, the correlation is not perfect, i.e. one having a correlation coefficient of 1.0. See correlation. • (verb, core´uh late) To compute, in order to quantify the complementary, or mutual statistical relationship between two variables. See correlation.

correlation • (noun, core uh lay´shun) Statistical technique that quantifies the extent to which two variables, quantified sets of observations, or groupings of numerical data change in conjunction with, or relative to each other. Correlation can be positive, in which case two variables change in the same polarity (both positive or both negative); or negative (one variable positive versus the other variable negative). Quantifications, or correlation coefficients range from perfect negative (–1.0) to perfect positive (+1.0).

correlative • (adjective, core rel´uh tiv) A correlative relationship does not necessarily mean that one variable causes the other–that would be a causal relationship. For example, the average hem length of ladies' skirts reputedly has a high positive correlation with current economic trends. (The higher the hem line–the higher the level of prosperity). See correlation. Manifestly, hem length of ladies' skirts cannot be manipulated to actually cause such economic fluctuations. However, it has been shown that these two variables do correlate thusly. (If this particular correlation were actually causal, we would probably experience a persistent vogue for shoulder length skirts. And, we would undoubtedly have a correlative trend toward open-necked trousers for men. "Let the good times roll.") On the other hand, tobacco use has been shown by rigorous clinical studies to actually cause health problems. Smoking and health have a causal relationship, not merely one that can be expressed by a correlation coefficient where no causal factors are in effect. • Zero (0) correlation means that no statistical relationship exists between two variables. The likelihood that two variables with zero (0) correlation would involve causality is certainly near zero (0) as well. However, the prospect of zero (0) correlation between any two things in the known universe also seems highly unlikely, as per the wisdom of the retired Jedi Knight (and obscure sound designer) Yoda (surname shrouded in mystery). See Star Wars (1977), Directed by George Lucas.

cps or CPS • Cycles per second (CPS) is an outmoded, albeit descriptive acronym for the unit of measurement for frequency. The acronym "cps" has been superseded by the term "hertz," abbreviated Hz (with no period), named in honor of Heinrich Hertz, a famous physicist and early radio experimenter. A one (1) kilohertz (1 kHz) signal has 1000 cycles per second (cps), meaning 1000 completions of any periodic waveform's cycle per each second, expressed as 1000 Hz. See (measurement unit) hertz. See frequency. Cf. pitch.

cross fade • A dynamic increase of the amplitude of one signal in tandem, i.e. coordinated with a proportional decrease in an associated signal, particularly between two audio signals. As one signal fades in (grows louder), the other fades out (grows quieter) proportionately. • A cross fade may be considered to be a negative correlation between respective amplitudes of two audio signals that exhibits a dynamically increasing negative correlation coefficient. As one signal is progressively amplified the other is progressively attenuated. A pan pot essentially cross fades a signal between left (L) and right (R) speakers in a stereo configuration. See correlation.

crosstalk • Unwanted sharing or intrusion of signal(s) among channels, i.e. interference of the data on one channel with another in a sound recording, information system, or communications medium. Cf. noise. Cf. interference. • Stereophonic long play (LP) vinyl records have significant crosstalk between left (L) and right (R) audio channels due to the mechanical (analog) nature of this audio playback system. In a modern digital sound system there is essentially no unintended crosstalk between stereophonic channels (L-R) or surround (5.1) channels.

cutoff frequency • The unique frequency that separates the pass band from the stop band on a cutoff filter. This frequency is aka the “half power point,” i.e. having a three decibel loss (–3 dB) with respect to those frequencies in the Pass Band that are not attenuated. See frequency, cutoff. See band, pass. See band, stop.

DAC • Digital to analog converter. See (converter) DAC.

damp • To progressively reduce the amplitude (size) of a vibration, waveform, or oscillation.

damping • Progressive reduction of energy or signal amplitude, resulting from loss of energy in a system. • A network or component within a system that brings about such a progressive loss of energy or signal amplitude. In electronics, any method used to reduce the amplitude of oscillations, using resistance or other energy absorbing components, is known as damping.

dB • Abbreviation for decibel, a power ratio measurement. See (measurement unit) decibel.

dB, zero • A 0 dB (zero decibel) signal level or power is any one of a number of standards used as a reference for comparison with other dB (decibel) powers or signal levels. Levels above 0 dB are expressed as positive (+) dB, and levels below 0 dB are negative (–) dB. For instance, –9 dB is –3 dB below –6dB, because –6 dB is closer to 0 dB. On the positive side of 0 dB, +9 dB is +3 dB above +6 dB. There are several established 0 dB references. In physics, 10^{-12} W / m^2 (read “ten to the minus twelve watts per square meter”), is an accepted 0 dB reference (also shown as 10^{-12} W m^{-2}). When this reference is used, 0 dB represents the power of a sound that is barely audible to the average human ear, a relatively small signal. This 0 dB reference is located near the lower end of the range of signals of interest. In audio engineering, there is a 0 dB reference located near the higher end of the range of signals of interest. That is, 0 dB $_{VU}$ on a recording console Volume Unit meter ≈ 0.775 volts, which is a large signal in audio engineering. Each 0 dB reference is established due to its relevance within the range of signal powers in its particular field. See (measurement unit) decibel.

DC • Direct current (DC) is electricity that flows in only one direction on a conductor, i.e. DC is a unipolar signal. DC is a steady flow that does not change polarity, being either positive (+) or negative (–), but not bipolar (±). DC is produced by rectifying (making unipolar) and integrating (smoothing, or low pass filtering) alternating current (AC); or by electrochemical action in cells that can constitute an electrical battery (DC source) when connected to each other. • In a virtual system, DC is represented by a constant whose magnitude (numerical value) and sign (±) remain the same until altered. See constant. See offset. See (polarity) unipolar. Cf. (polarity) bipolar. Cf. AC.

decay • In physics, the term that describes the time it takes a sound to die away, or fall to relative silence. This term is also used to indicate the half-life, or deterioration constant of a radioactive substance. Half-life is the specified time for a property to be diminished by one-half (0.5). • The last segment, or decay (D) of a two part attack & decay (AD) envelope signal. Also, the second, or decay (D) segment produced by an ADSR envelope generator. In the ADSR envelope generator, the decay (D) segment typically falls from some maximum positive level to another level (typically (S), or sustain) in a decay (D) time programmed by the user. In some designs, envelope generator signal segments such as decay (D) may be subdivided into several shorter segments that facilitate changing the curve of the composite decay (D) segment. See EG segment. See ADSR • In musical terms, "release" is preferred over "decay" to describe the timing of how an articulated sound ends. See release. Cf. attack.

decibel • See (measurement unit) decibel.

de facto • Situation whereby something exists in fact, whether by right or not. Something that exists as a fact due to its prevalence, not particularly due to official approval by some governing body or authority.

default • Initial status, level, number, or condition of a given parameter in a system. Defaults are typically defined by the designer to facilitate initial use of a device, or to optimize success using a system initially. Often, users can change or override defaults in order to achieve specific outcomes or to accomplish individual goals. For instance, the default gain factor in a two quadrant multiplier (signal controlled amplifier (SCA), or voltage controlled amplifier (VCA)) is typically zero (0). In this case, the VCA (SCA) can be used only to attenuate the carrier input signal, making the device useful for enveloping. (Envelope: the change of the intensity of a sound over time). See multiplier, two quadrant. Cf. multiplier, four quadrant. Cf. nominal.

delay • Interval of time an event, signal, or process is held in abeyance–temporarily caused to be inactive, inoperable, or not present prior to its actual introduction, appearance, or use.

delta • The amount of change (decrease or increase) in the magnitude (size) of a variable or function, particularly in relation to ongoing instantaneous changes of a signal. • In math, delta is a finite increment that may take on a specific value. The icon for delta is represented in math by an equilateral (equal sided) triangle with one side resting on the horizontal axis, with one apex pointing upward. Delta for the speed of sound relative to air temperature is 6 m/s per 10° Celsius. That is, the speed of sound in air changes approximately six meters per second per each change of ten degrees centigrade (Celsius). (The hotter–the faster) • In psychology, "delta" discrimination involves perception of the specific variable among several that have changed when two "trials" (instances) featuring several variables are presented serially (one after the other), e.g. (was there a change of color– or shape amongst similar symbols presented in this trial?)

demo • Abbreviation of demonstration, (pronounced dim´oh), a collection of recorded excerpts of sound designs, songs, arrangements, compositions, videos, film score cues, etc. that illustrate the capabilities of someone seeking work within the sonic arts community. • Demonstration of a device, computer software, or some process.

device, active • Hardware module or circuit that requires a power supply. The output of an active device, e.g. an oscillator, operational amplifier (op amp), noise generator, etc. may be considered to be the final outcome facilitated by that device's power supply. Cf. device, passive.

device, passive • Hardware module, circuit, or electronic component that requires no electrical power to perform its operations. The output of a passive electronic component (capacitor, resistor, diode, etc.) is a function of present and/or past input signals. • Some hardware fixed filters with toroids, or toroidal (doughnut shaped) coils as an integral part of their design are passive modules and require no electrical power. The input signal itself provides the (electronic) means to cause the module to operate appropriately.

*differential I/O • Jacks that provide both negative and positive signal polarities for the same signal. See jacks, differential input. See jacks, differential output.

digital • Representation of a physical phenomenon or signal using discrete rather than continuous means. • A system that generates, models, simulates, processes, represents, transmits, or transduces signals using quantized values intentionally constrained to be discrete, rather than continuous. Cf. analog.

digital signal • The ubiquitous digital signal represented produced by pulse code modulation (PCM) exhibits no changes of actual signal levels that directly correspond to level changes of the original analog signal it represents. The varying levels of an analog signal are not represented by corresponding, or analogous level changes in a PCM digitized signal, but are encoded to represent numbers. A PCM (sampled) signal has only two states or levels, “high” or “low,” and it is therefore a binary encoded signal. This is in contradistinction to the infinitely many different levels the original analog signal may exhibit. A continuous signal, i.e. one that is not encoded in steps, reduced resolution, or discrete levels. [Note: not all digital signals are binary; for instance, PAM (Pulse Amplitude Modulation) signals exhibit several different amplitudes, albeit still in discrete levels. • In contradistinction, amplitude changes of an analog signal are represented completely continuously, rather than as discrete, encoded steps as in pulse code modulation (PCM), aka sampling. An analog signal has levels that are analogous to the phenomenon it represents and features electrical correspondences with the levels of that phenomenon. Such correspondences are made with no discontinuities between successive levels over time. For example, a sound pressure wave and its equivalent electrical signal at the output of a microphone (mic) are both continuous, and each is analogous to, or an analog of the other. As sound pressure level (SPL) fluctuates continuously, the mic’s electrical output signal level mirrors these changes directly, proportionately, and continuously. Cf. analog signal.

diode • Electronic component that allows flow of electrical current in only one direction, i.e. current having only one polarity, plus or minus (±). The diode is a primary component in rectification of (bipolar) alternating current (AC) into (unipolar positive) direct current (DC) and is therefore a common component in DC power supplies. See rectifier.

direct box or DI box • Hardware input-output “stomp box” processor with an isolation transformer or other circuitry that facilitates input of signal(s) in a recording studio directly into the recording console, in lieu of using a microphone (mic) to pick up the sound of the instrument via the speaker through which it is played. A direct box allows connections of electric or electronic musical instruments directly to the console. Sometimes abbreviated DI for direct input (box). The direct box scales its input signal to the appropriate voltage level and provides appropriate impedance (resistance) matching between instrument (e.g. guitar) and recording console. An isolation transformer (when provided by the design) breaks the connection between the signal (chassis) ground and power line ground, thereby avoiding ground loops that can cause hum.

disk or disc • A low profile (“thin,” not thick), solid metallic cylindrical object that is also flat (without prominent protuberances or indentations), sometimes described as being similar to a “saucer” or “plate.” • Devices comprising one or more rapidly rotating metallic disks capable of storing data using electromagnetic or electro-optical means. Such (multiple) disks are found in a disk drive, a memory peripheral for a digital computer. • “Disk” is the preferred American spelling, particularly when referring to computer technology, except for “disc” as in compact disc (audio CD) or disc brakes for an automobile.

disk drive • A mechanism comprising motorized rotating disks featuring electromechanical or optoelectrical access elements known as “heads,” used for data storage of discrete signal levels (data) in a digital computer, DVD or CD player, laserdisc player, etc. See disk.

(display) frequency domain • Two dimensional graph or electronic representation of a signal, with frequency on the (x), or horizontal axis, and amplitude on the (y), or vertical axis. For example, a frequency response curve is shown in the frequency domain as a continuous line, using a graphic representation that features logarithmic (log) frequency on the horizontal (x) axis, and amplitude in dB on the vertical (y) axis. This is known as a semi-log, or log-linear graph. A frequency response curve represents the amplitude(s) of all possible frequencies in a given frequency band, typically from 20 Hz – 20 kHz when showing the overall frequency response of a piece of audio equipment. • The relative amplitudes of partials in a complex waveform are depicted individually by the relative heights of vertical lines in a line spectrum, another type of frequency domain display. A dynamic, electronic video presentation of individual partials in a signal is produced by a real time spectrum analyzer. Cf. spectrum, line.

(display) oscilloscope • An electronic device or virtual system that continuously and dynamically displays electrical signals in the time domain, with time on the horizontal (x) axis, and signal level on the vertical (y) axis. An oscilloscope represents sound, a longitudinal wave whose compressions and rarefactions of air pressure are difficult to show graphically, as a more-comprehensible transverse waveform. In this case, instantaneous sound pressure level (SPL) is displayed on the vertical (y) axis using a selected time base on the horizontal (x) axis. Several cycles of a waveform can be displayed, depending on the time base selected. The oscilloscope is used to view familiar geometric waveforms, e.g. sawtooth, square, triangle, sine. See wave, transverse. Cf. wave, longitudinal. See domain, time. Cf. domain, frequency.

(display) spectrum analyzer • An electronic or virtual system that analyzes a complex waveform, and displays the frequency, amplitude, and (in some cases) phase of each of the partials that constitute that waveform in the frequency domain. Frequency is depicted on the horizontal axis in either a linear or logarithmic (log) representation. See frequency representation, log. Amplitude is depicted as the height of the individual vertical line erected at the frequency of each partial. A spectrum analyzer may operate in real time, providing dynamic analysis and display. Line spectra provide a clear view of the partials that constitute a complex waveform. See spectrum, line. See partial.

(display) time domain • Two dimensional representation of a selected signal that depicts waveform, with amplitude changes on the vertical (y) axis (ordinate values), represented in time on the horizontal (x) axis (abscissa values). • The static, graphically based representation illustrates a waveform. e.g. geometric waveforms having amplitudes of partials that remain at the same level (so-called “static” waveforms). A dynamic, electronic presentation can be produced by an oscilloscope. See waveform. See abscissa. See ordinate.

display, waterfall • A two dimensional display that simulates a three dimensional view of signal characteristics derived from both frequency and time domains. Typically, signal amplitude (dB, voltage, SPL, etc.) is on the vertical (y) axis; frequency on the (x) axis; and time from "front to back" (bottom to top), or the converse, constituting a putative (z) temporal (time) axis. The waterfall display may be used to show how the amplitudes of constituent partials derived from a FFT (fast fourier transform) analysis of a complex wave evolve over time. That is, the envelope of each partial of a group of partials (complex waveform) is illustrated over time using a continuous line depicted on the (z) axis. The FFT resolves a waveform into its constituent individual frequency components (sine or cosine), and the added time dimension of the waterfall display retains some sense of the time domain. See time domain. See frequency domain. See FFT. See fourier analysis.

(display) sonogram • A two dimensional means of representing three signal characteristics, with frequency on the vertical (y) axis, time on the horizontal (x) axis, and signal intensity (z) or power indicated by contrast (light-dark) of the resulting graphic or electronic display. • Speech is often displayed as a sonogram, both graphically, and using electronic devices or computer programs that dynamically present a sonogram that evolves in real time.

distortion • In audio engineering, an unintended change of signal waveform that creates sound artifacts or anomalies that are usually unwanted. Distortion arises due to conditions present in audio system elements or equipment in use (e.g. clipping by an amplifier), in contradistinction to interference, which is caused by a signal or agency outside of, i.e. external to the audio system or electronic circuit in use (e.g. radio frequency interference (RFI)). Cf. (interference) etc. • The primary types of audio distortion involve signal amplitude, frequency, intermodulation, or phase. See distortion, etc. • In music synthesis, distortion techniques may be used to bring about intended changes in sound production or processing. For example, linear FM (frequency modulation) has often been characterized as a "distortion" synthesis technique. Deliberate introduction of noise and/or distortion can be valid in the sphere of artistic sound manipulation. See waveshaper, nonlinear.

distortion, amplitude • Unintended waveform changes due to nonlinear amplification of a signal. Amplitude distortion typically produces extra partials, particularly harmonics, not present at the input (e.g. due to signal clipping when using an amplifier). When such distortion occurs, the spectral content of the signal is changed; this type of distortion is aka harmonic distortion. See (distortion) clipping. See spectral.

(distortion) clipping • The specific form of amplitude distortion caused by overdriving an amplifier or output of any device, forcing that device to exceed its maximum rated output level. Clipping occurs due to the attempt to force an amplifier to produce an output level greater than the maximum output level afforded by its design. Clipping is quite evident when the top and/or bottom excursions of a simple input signal, e.g. sine wave are truncated, or "squared off" as a result of clipping. This scenario creates an output waveform with a "flat," horizontal line at some intermediate level less than the highest intended levels represented by the (unclipped) input waveform, as displayed in the time domain. • Clipping typically creates odd order, or more simply odd harmonics

(H), e.g. H3, H5, H7, etc. relative to a sine waveform input deemed to be the first harmonic (H1). When a complex waveform is clipped, many additional partials (sine waves) are created.

distortion, Doppler • See Doppler effect.

distortion, frequency • Coloration of a complex input signal due to a device's inability to pass or process frequency bands equitably. In this case, no new partials are produced, but the amplitudes of input partials are not processed proportionately. For instance, a device may exhibit poor low, mid, or high frequency response due to its inability to provide uniform gain changes in all input signal frequency bands. • When frequency distortion occurs, the device has a frequency response curve that is not flat. A flat frequency response curve approximates a horizontal line, indicating uniform device response to all frequencies. See frequency response, flat. See frequency response.

distortion, harmonic • Particular type of amplitude distortion that creates new partials that are harmonic to some input partial(s). See distortion, amplitude.

distortion, intermodulation • Extra partials produced by input partials that interact to produce summation and/or difference frequencies not originally in the input signal or partials unrelated to input partials. In particular, use of a single loudspeaker or driver to produce all the frequencies of a complex program signal (e.g. music) likely produces intermodulation (IM) distortion. Speaker cabinets with individual drivers specifically designed to produce frequencies in a specific limited band (e.g. woofer, tweeter) tend to produce less intermodulation distortion.

distortion, nonlinear • Extra partials created due to a transfer function of a device that is not a straight line (linear). Clipping is a severe form of nonlinear distortion. See (distortion), clipping. • Nonlinear wave shaping in sound synthesis is a form of distortion that is intended, and various devices or modules are designed having variable nonlinear transfer function(s) between input and output. See function, transfer.

distortion, phase • Shift(s) of phase, caused by time delays of particular partials in the input signal, constitute phase distortion. The human ear is reasonably insensitive to phase changes, so this type of distortion is subtle, but can be shown to have a real effect in a good listening environment. Also, the ear is evidently more sensitive to phase anomalies at lower frequencies than higher.

distortion synthesis • Any of a variety of means, particularly modulation techniques, e.g. FM, AM, etc. that produce partials not originally present in the carrier (C) and/or modulation (M) signals. For instance, the sidebands produced by classic amplitude modulation (AM), balanced amplitude modulation (BAM), or linear frequency modulation (FM) may be considered to be "distortions" of the original carrier waveform. In this context, "distortion" is intentional, and is not considered to be production of "unwanted" partials. See distortion, etc.

domain, frequency • Two dimensional representation of a signal, with frequency on the (x), or horizontal axis, and amplitude(s) of partials on the (y), or vertical axis . The relative amplitudes of partials in a complex waveform are depicted by the relative heights of vertical lines in a line spectrum, which is a frequency domain display. • A dynamic, electronic presentation of individual partials in a signal is produced by a real-time spectrum analyzer. See spectrum, line.

domain, time • Two dimensional representation of a selected signal that depicts its waveform, with amplitude on the vertical (y) axis (ordinate values), and changes over time shown on the horizontal (x) axis (abscissa values). • A static, graphic representation illustrates a waveform, a time domain display. A dynamic, electronic presentation is produced by an oscilloscope to which the waveform is connected. See waveform.

Doppler effect • The perception of the objective, dynamic change of a wave's frequency due to relative motion of wave source and listener. That is, Doppler effect is not a perceptual artifact, but an objective change of frequency. • Doppler effect is heard as a progressive pitch change that becomes higher as an audio point source (e.g. railroad locomotive horn, police siren) and listener move toward each other (converge), and lower as they move away from each other (diverge). • In astronomy, Doppler effect is aka Doppler shift, and involves light waves. Doppler shift is called "blue shift" as light wave and observer move progressively closer (converge), and "red shift" as they diverge. This dynamic wave frequency change is named after Austrian physicist Christian Doppler (1803–1853).

DSP • Digital signal processing (DSP) involves use of a digital computer or dedicated digital circuitry to manipulate virtual, i.e. discrete representations, or numerically-based signal(s) mathematically. See virtual. See virtual reality.

duration • Timing of, or time an event or a portion of an event takes to complete.

duty cycle • See waveform duty cycle. • The average time a device is rated to operate prior to failure.

dynamic • Relating to elements, forces, signals, or parameter settings that produce motion or change. Moving, or having motion. Cf. static.

dynamics • The range of loudness variations in standard music notation, expressed by the Italianate musical markings pp for pianissimo (very soft) through ff for fortissimo (very loud), particularly in printed music. Dynamics are an important part of music, whether notated or not. Some urtext editions are printed without many such performance marks, which may include tempo indications, marks for phrasing, dynamics, and other marks or words (e.g. con sordino, play using a mute).

dynamic range • The span of loudness levels (LL) that occur in a particular piece of music, bracketed by the extremes (least vs. most powerful) of those levels. The dynamic range of perceived loudness in a musical composition is expressed arbitrarily using

"Italianate" subjective dynamic markings. See dynamics. • The intensity range over which an audio system can operate without distorting, predicated on a limited bandwidth (frequency span) and other design specifications. The dynamic range of electronic equipment is described non-arbitrarily, i.e. objectively, primarily by inter-related measurements of signal frequency (Hz) and amplitude (dB), given a specific percentage of distortion.

echo • Reflection of a sound wave, heard later in time relative to a more direct path of that sound wave to the listener. Multiple echoes make up reverberation, although "echo" is usually taken to involve longer delay time(s) than reverb. • Music halls are designed to optimize the density (number in a unit of time) and total reverberation time of the many echoes caused by multiple reflections of sound waves. Such venues are specifically designed to facilitate multiple echoes, having an optimal reverb time. See reverberation. See (wave propagation) reflection.

editor • Computer software that facilitates programming of sound parameters and/or musical elements, often used in conjunction with "librarian" software that stores "patches" that comprise sound parameters, etc. See patch.

e.g. • Abbreviation for "example given," after which one or more examples among many possibilities are provided.

EG • Generic non-pronounceable acronym for an envelope generator (EG) module. See generator, envelope. See ADSR.

EG segment • Distinct part, i.e. discernible subdivision of the overall signal produced by an envelope generator (EG). EG signal segments have been named using letters such as A, D, S, R for attack, decay, sustain, release. See ADSR. In general, an EG segment is defined, and depicted graphically, as a line or curve between two levels traversed in a single time. Determination of these two (2) levels and single (1) time is a general model for the programmable parameters that define such envelope generator segment(s). For example, level one (L1), level two (L2), and time one (T1) would comprise a single EG segment. This indicates that the given EG segment starts at level one (L1) and moves to level two (L2) in a time one (T1). (Numbers here act as names only. L1, L2, and T1 parameters as indicated would have numerical values that could be programmed independently). In some cases the very first, or initial level is named "L0" (level zero). The next level would then be designated as "L1." Additionally, "L0" might also be used as the final level to which the final segment falls, representing zero "0" signal amplitude. • An envelope generator (EG) typically produces several (EG) segments serially, with no discontinuities from beginning to end of the overall output signal. Therefore, each segment's ending level is the same as the following segment's starting level. That is, if L1 and L2 are levels for the first segment, then L2 and L3 will be levels for the second segment, and so forth. Conversely, a given segment's starting level is the same as the previous segment's ending level (very first EG segment excepted for obvious reason). The starting level for the first EG segment (attack) is typically zero (0)), as is the ending level for the last EG segment (release). See generator, envelope.

electroacoustic • Dealing with generation, transmission, and transduction of electronic signals into sounds, and sounds into signals. • A genre of music defined by such means of making music, using electroacoustic musical instruments or device(s) as a necessary or sufficient means of production. In this context, "electroacoustic music" does not imply any particular musical genre or style, only the means and processes for making such music. See elektronische music. See musique concrète.

elektronische musik • Electronic music, particularly from the early studio founded by Herbert Eimert, Robert Beyer, and Werner Meyer-Eppler in Köln, Germany in the early 'fifties at the NWDR (Northwest Deutsch Rundfunk) radio station. The earliest compositions by the founders used electronic instruments such as Harald Bode's Melochord and Friedrich Trautwein's Monochord as sound generators. Soon thereafter, early elektronische musik compositions of Karlheinz Stockhausen and others realized at the Köln studio featured sine waveforms as the primary sonic resource, particularly in lieu of analog recordings of acoustically-generated sounds as per musique concrète, which originated in Paris, France circa 1946–1948. Electronic musical instruments per se were abandoned in favor of making the tape recorder the favored medium for composition. The central interests of early composers of electronic music at the NWDR sprang not so much from a focus on creating new sonic resources, but primarily from an interest in controlling sound to better achieve complete serialization of the tone rows in dodecaphonic (12 tone), or serial music. The "house organ" publication from the early Köln studio had the revealing name Die Reihe, translated as The Row. The first edition was subtitled "Information über serielle Musik" ("information on serial music"), clearly indicating the primary interest in dodecaphonic (12 tone) music, rather than electronic sounds as such. Cf. musique concrète. • The early genre distinctions between musique concrète (Paris, France ca. 1948), featuring manipulations of recorded sounds; and elektronische musik, (nominally featuring sine wave oscillators at the "first" (ca. 1950) electronic music studio in Köln, Germany), are essentially only of historical interest, as techniques derived from both modalities of composition are in widespread use today.

enharmonic • An alternate "spelling" or representation of note(s) in a musical structure (chord, melody, passage) that would have the same or essentially "equivalent" pitch if expressed in a different tonal center, or "key," particularly when played on a keyboard tuned to the equitempered pitch system. For example, d-sharp, e-flat, and f-double flat are enharmonic "spellings" of the same note when notated, and therefore share the same key on a piano keyboard. Enharmonic spellings become necessary, and particularly useful when notating highly chromatic music, with passages involving many different and/or evolving "keys," or tonal centers. • The term "enharmonic" should not to be confused with inharmonic, meaning a single given partial that is part of a complex waveform, that is not harmonic to other partial(s) or the fundamental of that complex waveform. See (partial) inharmonic. See (partial) harmonic.

envelope • The dynamic, infrasonic (below audible frequencies), typically aperiodic changes of an articulated sound's intensity over time. In the case of electronically produced sound, an envelope results from changing the amplitude of a monitored audio signal over time. • Strictly speaking, envelope describes a sonic characteristic: the dynamic changes of the intensity of a sound over time. Such changes of signal intensity correlate with loudness changes in cases where the signal is audible. However, due to persistent usage, the concept of "envelope" has been generalized to include dynamic, infrasonic, aperiodic changes of signal characteristics (and possible related sonic attributes) other than intensity, e.g. spectrum "envelope" (timbre changes), and frequency "envelope" (pitch changes). See dynamic. See infrasonic. See aperiodic. • In musical terms, "envelope" describes variations in the articulation, or shorter-term loudness changes of sound over time. Longer-term changes of loudness over time in music are generally referred to as dynamics. See dynamics. See generator, envelope. See ADSR. See EG segment. • Even though the envelope signal produced by an envelope generator (EG) is aperiodic, it is not a random signal. All random signals are aperiodic, but not all aperiodic signals are random. Cf. noise.

envelope follower • Processor that rectifies, i.e. changes a bipolar (±) signal to a unipolar positive (+) signal, and dynamically integrates (smoothes), or averages the power of this rectified signal using a specific time constant (as provided by a low pass filter). Envelope follower (EF) output comprises a unipolar positive, infrasonic (lower than audible frequency span), aperiodic, fluctuating direct current (DC) signal or its virtual equivalent. This signal is similar to the output of an envelope generator (EG). But, envelope follower (EF) output is not programmed by the user by providing values for various levels and times. See envelope generator. See EG segment. Envelope follower output is produced dynamically and continuously and is directly proportional to the level of the alternating current (AC) input signal, typically an audio signal. • To extract the envelope of an input signal, averaging its instantaneous amplitude changes is required. If an envelope follower (EF) followed every amplitude change at its input literally, then the EF output signal would simply duplicate the EF input signal. The degree of smoothing, i.e. amount of averaging, is programmable in some designs, and may be expressed either as a time constant or the cutoff frequency of a low pass filter (LPF). A low pass filter has also been called a "lag processor" or more generically, an integrator. As cutoff frequency is lowered, progressive smoothing of the more-rapid (higher slew rate) input level fluctuations is provided. • A standard volume unit (VU) meter functions somewhat like an envelope follower, displaying the effective (averaged) input signal level, rather than every instantaneous plus and minus (±) excursion of input signal amplitude relative to zero (0). That is, even though a sine waveform tuned to 440 Hz makes 440 excursions through positive and negative polarities per second, the needle of the VU meter does not move up (+) and down (–) 440 times per second. The VU meter displays the averaged level of such a sine wave, which appears as a single steady value. See envelope. See rectifier. See diode.

envelope generator • An aperiodic signal generator capable of producing, when triggered or gated, a unipolar positive, aperiodic, non-random, infrasonic, segmented, timed signal. In terms of analog modules, an envelope generator produces fluctuating direct current (DC). The ADSR variant of envelope generator (EG) was defined conceptually by Vladimir Ussachevsky, of Columbia-Princeton Electronic Music Center fame, and first built (model 911) by Robert A. Moog on his eponymous modular 900

Series analog voltage controlled synthesizer systems. Modern envelope generators typically have a given number of segments (e.g. A-D-R of the commonplace ADSR envelope generator), each of which moves between two (2) programmable or predetermined levels during one (1) programmable time. There have been “rate-based” envelope generator designs (e.g. Yamaha Model DX7), in contradistinction to the ubiquitous time-based designs seen today, but this rate-based type has fallen out of favor (thankfully). See ADSR.

envelope signal • A unipolar positive, aperiodic, non-random, infrasonic, segmented, timed signal. Individual segments of an envelope signal such as A, D, and R (attack, decay, release) typically comprise two levels, and a time taken to move from one level to the other. The S (sustain) segment typically stays at the same (programmed) level as long as the signal connected to the Gate Input of the envelope generator is positive in polarity. See ADSR. In analog terms, an envelope signal may be thought of as fluctuating direct current (DC). Cf. envelope. Cf. envelope generator.

EQ • In modern terms, equalization (EQ) involves filter(s) that alter tonal balance, i.e. relative strengths of various frequency bands in an audio signal or mix of audio signals. These filters are of several varieties: shelf, parametric, etc. Some EQ capability resides within each channel strip of a mixing console. See mixing console. See equalizer, shelving. See filter, high pass. See filter, low pass. Other equalizers may be mounted in racks within the studio and are accessed remotely using the send and return busses on a mixing console. See equalizer, graphic. See equalizer, parametric. • At its inception, equalization (EQ) restored the intensity of telephonic signal(s) communicated over long distances and had little or nothing to do with user programmable changes of tonal balance.

equalizer, graphic • Broadband processor comprising many band pass filters (BPF) tuned to standardized center frequencies, typically at 1 octave, ½ octave, or 1/3 octave intervals, to create overlapping “narrow” bands within the bandwidth of potentially audible frequencies. The output level of each BPF, or “section” of the resulting equalizer is determined by a single slide-pot (potentiometer). These slide-pots are oriented vertically and arranged side by side in close proximity so as to “graphically” represent the approximate frequency response curve provided by their collective settings. The tonal balance provided by a graphic EQ can be determined at a glance by looking at the relative positions of the slide-pots, each of which provides the relative strength of its frequency band. Each slide-pot in a graphic equalizer (EQ) typically provides a choice of either “boost” (amplification) or “cut” (attenuation) of its associated frequency band, perhaps over a ± 12 dB or greater span.

equalizer, parametric • Broadband processor that provides several band pass filters (BPF) independently tunable in center frequency, bandwidth (aka as “Q”), and boost or cut (e.g. ± 15 dB) characteristic. In most designs, tuning of the center frequency of a particular BPF has frequency limits, although the restricted center frequencies of the several band pass filters may overlap significantly. A parametric equalizer (EQ) can effectively deal with “local” frequency response problems or provide opportunities in processing a signal or mix, e.g. elimination of hum at 60 Hz, creation of formants at specific center frequencies, etc. See (interference) hum. See formant.

equalizer, shelving • Filter(s) with a flat response either above or below a particular fixed frequency, providing boost (amplification) or cut (attenuation) of the frequency band that is not part of the flat response. A "high" shelving filter has a flat frequency response below a fixed high frequency, and provides boost (amplification) or cut (attenuation) only above that fixed frequency. A "low" shelving filter has a flat frequency response above a fixed low frequency, and provides boost (amplification) or cut (attenuation) only below that fixed frequency. The "shelf frequencies" or "shelf" comprises those frequencies that are either amplified or attenuated.

equal loudness curves • See loudness curves, equal.

ergonomics • The science and art of determining optimal man-machine interfaces or operating conditions, aka human engineering or biotechnology. The study of workplace conditions that takes into account worker productivity in conjunction with human effort, comfort, and efficiency. Ergonomics necessarily involves an understanding of human anatomy, e.g. range of motion, and sensory perception.

et al • Abbreviation for the Latin "et alii," literally "and other persons." Most linguistic authorities believe use of this abbreviation should be restricted to refer only to people, not to things. Cf. etc.

etc. • Latin for et cetera (abbreviated "etc."). Indication that a given list of things is incomplete; that there are other unspecified items of possible interest. From the Latin et meaning "and," and cetera meaning "the rest." The terms "et al" and "et cetera" (etc.) are now so commonplace that it is no longer necessary to italicize them in print, as per the former convention. Cf. et al.

factor • Either of at least two numbers, which when multiplied create a product, which is a resulting number. The numbers 2, 3, and 5 are all factors of their resulting product 30, because two times three times five equals thirty (2 x 3 x 5 = 30). • Musical intervals represent ratios that result when multiplying a factor such as the frequency of a given note, e.g. 440 Hz–which is aka the note "A4," by a second factor (e.g. 1.05946). The result (product) of this arithmetic provides the second (upper) note of the interval. In this example, the factor 440 is multiplied by the factor 1.05946 to produce a new frequency of 466.2 Hz, that is approximately an equitempered half step, or one semitone higher than the original frequency, representing the new note A-sharp (A#4). The ratio of 1.05946:1 therefore approximates the frequency ratio of an equitempered half step. Multiplication of any frequency by the reciprocal (1/1.05946) of this ratio, which is 0.9438 produces a frequency one semitone lower. The factor 2.0 produces a resulting frequency (product) one octave higher. Multiply by the reciprocal of 2.0, which is ½ (0.5) to produce a new frequency one octave lower. A factor need not be a whole number, such as 2, or even a rational number, i.e. one that is expressed by a ratio made up exclusively of integers (whole numbers), e.g. 3:2 (Perfect fifth, or P5). As noted, the factor for an equitempered musical semitone (half step), which is approximately 1.05946 is an irrational number, aka the twelfth (12th) root of two (2). If one starts with any given frequency and successively multiplies that frequency (and then each resulting product) by this same factor (1.05946) twelve times, the musical interval, or ratio of approximately an octave (2:1) will result. The procedure illustrates that there are twelve (12) equally-spaced half steps in an octave (2:1) of the (equitempered) musical scale. See interval, musical. See ratio. See factor. See product. See integral. See number, rational. See number, irrational. You just have to love irrational numbers–they're so goofy!

fader • A slide potentiometer (pot), particularly one of the audio signal attenuators most prominent on each channel strip on an audio recording console. A fade is progressive attenuation, or reduction of sound over time to silence, which is done using faders on a recording console or mixer. The term "fader" properly describes an attenuator used exclusively to reduce the amplitude of audio (A) signals. A potentiometer (pot) used to reduce the amplitude of signals other than audio, e.g. control (C) signals, is properly called an "attenuator," the more general term. That is, all faders are attenuators, but not all attenuators are faders. And, not all people agree that it is critical to make such distinctions. (More's the pity). See attenuator. See potentiometer.

feedback • The routing of a portion of the output of a system, circuit, or module "back" into its own input to effect a change in performance, capability, or particularly to alter phase or frequency response. Filters feed back specifically designated phase-shifted frequency bands that are mixed with the original input signal to effect either or both attenuation (cut) or amplification (boost) of selected frequency band(s). • The howling "feedback" frequencies from a loudspeaker or public address (PA) system are due to positive feedback (aka constructive interference), caused by acoustic coupling of microphonic or musical instrument vibrations that are "fed back," or reinforced by room tone, i.e. resonant frequencies caused by the shape and size of the room. Room tone causes one form of feedback, which explains why specific notes are easier to "sustain" when playing an electric guitar at a high intensity level in a particular venue. Such "favored" notes are located at or near the resonant frequencies of the room. See room tone.

FFT • Fast fourier transform (FFT) is a technique of fourier analysis that is less computationally intensive, therefore faster in its execution. See fourier analysis.

file • Data, programs, information, etc. that reside in memory locations in a computer under a single identifying header which we refer to by "file name," e.g. named sound (audio) files.

filter • Processor that alters amplitudes (or in some cases only phase relationships) of partials in selected frequency bands of the processed input waveform. The term "filter" implies removing something, but some filters can boost (amplify) partials in selected frequency bands, as well as cut (attenuate) them. See filter, etc. See equalizer. • An ideal all pass filter (APF) does not change the amplitude of any partial passing through it, but does change the phases of partials in selected band(s). See filter, all pass. See phase.

filter, all pass • Processor with electronic circuitry or DSP algorithms that introduces a specific amount of phase shift (time delay or alteration) of partials in selected frequency bands in the processed signal, without significantly changing the amplitude of any partial–hence the designation “all pass.” An all pass network is not a self-contained module typically available to users, but several all pass networks are used in the design of a phaser, a familiar processor. • Filters process signals in the time domain to create boosted peaks, due to constructive interference; and/or attenuated valleys, due to destructive interference, in output frequency response. These changes in frequency response are due to feedback of phase shifted partials that are summed with the filter’s unprocessed, “dry” input signal. See feedback. • A phaser requires a separate all pass network per each peak or valley created in the output frequency response. The all pass networks in a phaser are aka “stages,” e.g. in a 12 stage phaser. • In contradistinction, a flanger, whose aural effects may be confused with a phaser, sums its input signal with a single dynamically delayed version of that signal. A flanger creates a comb filter frequency response that has many peaks and deep notches or valleys (nulls), using a single internal delay. See flanger. See filter, comb.

filter, all pole • Terminology used by DSP engineers to describe band pass filters used in linear predictive coding (LPC), a subtractive analysis-resynthesis technique used in speech synthesis, etc. In general, a filter pole is a resonant frequency band, i.e. a peak or formant in a spectrum. A filter that creates several smooth peaks is called an all pole filter. Cf. filter, all zero. See formant.

filter, all zero • Terminology used by DSP engineers to describe band reject filters used in linear predictive coding (LPC), a subtractive analysis/resynthesis technique used in speech synthesis, etc. In general, a filter zero is a notch frequency, i.e. a null, or valley in a spectrum. A filter that creates several notches is called an all zero filter.

filter, anti-aliasing • Processor positioned in front of the input of the analog to digital converter (ADC) in a well designed sampling system. Placed before the sampler’s ADC input, this brick wall (very steep slope) low pass filter removes partials that exceed the Nyquist frequency. These partials would otherwise become aliases, partials with erroneous frequencies in the output, due to sampling error. See alias. See aliasing. See frequency, Nyquist. See filter, brick wall. Cf. filter, smoothing.

filter, band pass • Cutoff filter that passes frequencies around a tunable center frequency f_{ctr}, (spoken “f sub center”), which lies between two implied cutoff frequencies that define the pass band per se. The band pass filter (BPF) has a bandwidth defined by the distance between the two cutoff frequencies (half power points) that surround the center frequency. See half power point. See frequency, cutoff. See filter slope. Cf. filter, band reject. • A BPF can be created by connecting a high pass filter (HPF) and a low pass filter (LPF) in series, and tuning HPF cutoff frequency lower than LPF cutoff frequency. Actual BPF designs do not necessarily implement such a configuration.

filter, band reject • Cutoff filter that stops, or rejects frequencies around a variable center frequency f_{ctr}, (spoken as "f sub center"), which lies between two implied cutoff frequencies. The band reject filter (BRF) has a bandwidth defined by the distance between the two cutoff frequencies that flank its center frequency. See frequency, cutoff. See filter slope. • A BRF can be created by connecting a high pass filter (HPF) and a low pass filter (LPF) in parallel, and tuning HPF cutoff frequency higher than LPF cutoff frequency. The outputs of the two filters are summed (mixed). Actual band reject filter (BRF) designs do not necessarily implement such a configuration, but comprise other circuit designs. • A BRF is aka a "band stop filter (BSF)."

filter, band stop • See filter, band reject. Cf. filter, band pass.

filter, brick wall • Low pass filters (LPF) used in a pulse code modulation (PCM) sampling system at both input and output. "Brick wall" describes the characteristic steep slope (–90 dB/Octave or greater) of a LPF that severely attenuates or "cuts off" partials above its cutoff frequency. This cutoff frequency is fixed by the designer to lie slightly above the Nyquist frequency, to ensure that aliasing does not occur at sampler input, and smoothing of the reconstructed analog signal does occur at sampler output. The brick wall filter in front of the input is aka an anti-aliasing filter; and the one after the output is aka a smoothing filter. See filter, anti-aliasing. See filter, smoothing. See aliasing. See frequency, Nyquist. See filter, cutoff. See filter, low pass.

filter, comb • Frequency response curve that looks like a hair comb when depicted graphically in a linear frequency presentation. The peaks and valleys of a comb filter may be created using feedback of a single delayed version of the processed signal summed with the unprocessed, or "dry" input signal. See feedback. • Because a comb filter frequency response occurs when any signal is mixed with a delayed version of itself, a "comb filter" is not necessarily a freestanding filter module that can be programmed by a user. For example, a flanger mixes a single, but variably delayed version of its input signal with the unprocessed input signal, to create various comb filter frequency response curves that can be changed dynamically using a variable delay time. See flanger. Cf. phaser. • The first peak of a comb filter is located at the frequency that corresponds to the period (T) of the delay time presently in use. A delay with a period (T) of 1/1000 second (one millisecond, or 0.001 second) corresponds to a frequency (f) of 1000 Hz, which then comprises the first (i.e. lowest) frequency peak of the comb filter produced by such a delay time. The first, i.e. lowest frequency valley, or null is ½ the frequency of the first peak, 500 Hz in this example. Peaks at higher frequencies occur at integer (whole number) multiples of the first peak, e.g. 2000 Hz, 3000 Hz, 4000 Hz, etc. Valleys at higher frequencies appear equidistant between peaks, e.g. 1500 Hz, 2500 Hz, 3500 Hz, etc. See frequency. See period. See relationship, inverse.

filter, cutoff • Processor that does what its name implies: a cutoff filter "cut offs," or attenuates partials in the input signal, in selected frequency band(s). Attenuated partials lie primarily in one or more stop band(s). Complementary pass band(s) have partials that are not "significantly" attenuated. That is, no partial in a pass band is attenuated more than three decibels (–3 dB), by definition. The boundary between stop band and pass band is defined by a single cutoff frequency, represented f_c (spoken "f sub c") in low pass and high pass filters. Or, by a single center frequency, represented f_{ctr} (spoken "f sub center") in band pass and band reject filters. A center frequency is flanked (surrounded) by two implied cutoff frequencies at –3 dB with respect to frequencies not attenuated. See filter cutoff frequency. See filter slope. See half power point.

filter, fixed • Filter with various pre-determined band pass sections, each of whose "Q" (bandwidth) and center frequency are not alterable. Bands in a fixed filter offer boost (+) or cut (–) only.

filter, formant • A fixed filter whose band pass section center frequencies are selected to specifically replicate formants in musical instruments or the human voice. See formant.

filter, high pass • Cutoff filter that cuts off, or attenuates frequencies below its cutoff frequency, and passes those above. The high pass filter (HPF) is aka a "low cut" filter. See filter slope. See filter, cutoff.

filter, low pass • Cutoff filter that cuts off, or attenuates frequencies above its cutoff frequency, and passes those below. The low pass filter (LPF) is aka a "high cut" filter. See filter slope. See filter, cutoff.

filter, notch • Cutoff filter that functions like a band reject filter (BRF) that has a very narrow pass band, with extremely steep filter slopes. See filter, band reject. See filter slope.

filter, resonant • A band pass filter featuring one or more sections capable of "resonating," or boosting specific frequency band(s). See filter, band pass.

filter, rumble • Cutoff filter that attenuates or removes frequencies below a fixed, very low cutoff frequency typically dictated by the designer, originally so-named because it reduced the "rumble," particularly very low frequency and putatively infrasonic noise produced by a vinyl long play (LP) record turntable. This filter might be switched in/out of the audio circuitry of the preamplifier that receives phono (LP turntable) output. See infrasonic. • The slope of a rumble filter is generally moderate to quite steep. • A type of high pass filter.

filter, scratch • Cutoff filter that attenuates or removes frequencies above a fixed, high cutoff frequency typically determined by the designer, so-named because it reduces the "scratch," or surface noise generated by vinyl long play (LP) records. This filter typically is switched in/out of the audio circuitry of the preamplifier that receives phono (LP turntable) or analog tape output. • The slope of a scratch filter is typically not extremely steep, so as to strike a balance between reducing high frequency noise, while not robbing the musical program of desirable highs. • A type of low pass filter.

filter, shelving • A filter that boosts/cuts frequencies either above or below a specific fixed frequency. See equalizer, shelving.

filter, smoothing • Processor placed after the output of the digital to analog converter (DAC) in a sampling system. This brick wall (–90 dB/Octave or steeper slope) low pass analog filter (LPF) removes high frequency artifacts inherent to the sampling process, and smoothes the series of amplitude pulses produced by the DAC. This smoothing filter facilitates reconstruction of the analog signal originally sampled. • The DAC smoothing filter cannot remove aliases caused by failure to use a complementary anti-aliasing filter in front of the input of the analog to digital converter (ADC). That is, after an analog signal has been digitized by the ADC, the sampling system cannot distinguish between partials sampled legitimately, and aliases produced by violating the Nyquist limit, since both kinds of partials have frequencies that typically lie in the pass band of the smoothing filter, i.e. below the brick wall LPF cutoff frequency. • Cf. filter, anti-aliasing. See frequency, Nyquist.

filter, voltage controlled • Filter with one or more parameters (e.g. cutoff frequency, resonance) whose values, or operating points, can be controlled by the voltages (or virtual equivalents) connected to associated control input(s) that are ported to an internal summing node, allowing input of several external control signals. For example, VCF cutoff frequency can be changed dynamically to follow the rate(s), shape(s), and polarit(ies) of the sum of all control signal(s) connected. A voltage controlled filter (VCF) typically provides, at minimum, dynamic control of cutoff frequency. Some implementations also offer dynamic control over resonance, aka “Q.” • In virtual systems, use of the term “voltage” may become vestigial (no longer useful), so signal controlled filter (SCF) may be a more useful designation.

filter cutoff frequency • Frequenc(ies) that determine the boundar(ies) between pass band(s) and stop band(s) in a cutoff filter. This boundary for low pass and high pass filters is determined by a single cutoff frequency. The boundaries for band pass and band reject filters is determined by two cutoff frequencies that flank a single center frequency. • Technically, cutoff frequency is the unique frequency that defines the boundary between stop and pass band(s). Cutoff frequency intensity, or power is attenuated by one half (½), which is –3 dB relative to the pass band frequencies that are not attenuated. Cutoff frequency is aka the half power point, as a 3 dB loss (–3 dB) is a halving (1/2) of power. See filter slope. See frequency, cutoff. See half power point.

filter Q • See filter resonance.

filter resonance • Feedback loop from filter output to input that increases energy at the cutoff frequency. Also known as “Q,” or “regeneration,” or “emphasis,” or “feedback.” Particularly associated with low pass and high pass filters whose slope can be changed using this “resonance” control. See resonance.

filter slope • Representation of progressively greater attenuation of stop band frequencies more distant from the cutoff frequency (half power point), e.g. in a cutoff filter. The transition between pass band and stop band in a cutoff filter is not abrupt, from zero attenuation to complete attenuation. Attenuation is progressive, represented by a slope: the more distant a stop band partial is from the cutoff frequency, the greater the attenuation of that partial's amplitude. • Filter slope is quantified in decibels per octave, e.g. –24 dB/Octave (spoken as "minus 24 decibels per octave.") Slope is assumed to involve a loss (–), i.e. progressive attenuation. Each 6 dB of attenuation is created by a single filter section, aka a "pole." A four pole low pass filter has four poles connected in series, resulting in a –24 dB/Octave slope. The more poles (filter sections) connected in series, the steeper the slope. See cascade.

fixed • Description of a temporarily unchanging (static) signal level or numerical value, particularly in relationship to parameters on a module or values in a sound synthesis system. Fixed values, aka "constants" in a virtual system, are typically user-programmable, but remain "fixed" (at the same value) until changed; for example, bias signals or virtual constants (e.g. coarse and fine tune frequency controls on an oscillator, cutoff frequency on a filter). A fixed signal or parameter value does not fluctuate of its own accord. However, such a programmed value is not immutable (unchangeable). A fixed value, sometimes referred to as a "static" setting, may typically be changed by the user. Cf. dynamic. • The term static also refers to a type of crackling noise, or interference in an electronic system. See static.

flanger • Processor that mixes a delayed replication (copy) of its input signal with the original untreated, or "dry" signal, thereby creating a so-called "comb filter" frequency response. This delay time may be changed by an internal low frequency oscillator (LFO) in order to dynamically move, or "tune" the frequency peaks and valleys in the comb filter frequency response that is produced. This effect is known as "flanging." See filter, comb. Cf. phaser.

flow chart • An iconic representation of the steps in an operation, or the sequence of steps in a process, particularly one which deals with software commands or operations in a digital computer. A flow chart typically has icons for logical or mathematical operations such as "compare," "increment/decrement," and branching "if, then, else" decision nodes that route steps or alter the sequence of steps within software algorithm(s) written for a computer.

FM • Frequency modulation (FM), a synthesis technique and engine, aka linear FM. In classic linear FM, carrier (C) and modulation (M) signals are sine waves. When both C and M are in the audio frequency range, modulation of C frequency by M frequency produces pairs of sidebands (sine waves) at (±) integer multiples of the specific M frequency. That is, the carrier and sidebands appear at frequencies: $c \pm km$ where c = carrier frequency, m = modulation frequency, and k = 0, 1, 2, 3, 4, and so forth, representing the carrier (0) and sideband pairs by their order of appearance (1, 2, 3, 4, etc.) as modulation depth is increased. See FM modulation index. • FM can produce partial(s) with negative, as well as positive frequency–and negative, as well as positive amplitude. The polarity of a "negative" partial is opposite (inverted) from that of a "positive" partial. See frequency, negative. See amplitude, negative. See FM etc.

FM carrier • In classic FM, a sine wave oscillator whose frequency is modulated periodically in order to produce sideband pairs of sine waves. Some implementations allow selection of carrier and modulator waveforms other than the sine wave.

FM modulation index • Depth of modulation, or carrier frequency deviation, expressed by the formula: modulation index = carrier frequency deviation / modulating frequency (carrier frequency deviation divided by modulating frequency). • In practice, few implementations display the actual modulation index, a number that typically ranges from zero (0) to approximately twenty (20). Rather, some arbitrary calibration (e.g. 0–99) of the output of the modulator (M) provides an undefined correlate of the actual modulation index. As per the modulation index formula, an increase in modulator output level causes an increase in carrier frequency deviation, and hence a larger index of modulation.

FM modulator • Periodic generator used to modulate (change) the frequency of a periodic carrier wave in order to produce complex waves.

FM operator • Periodic generator in linear FM, homologous to a sine wave oscillator whose output amplitude is scaled dynamically using an associated envelope generator. Linear FM's periodic generator.

FM potential spectrum • Spectrum of FM sidebands (partials) generated when the depth of modulation, or FM index of modulation is theoretically maximized. When the modulation index is zero (0), no sidebands are produced, hence the concept of a spectrum that "potentially" could be produced using a non-zero modulation index. See FM modulation index. • The frequencies of the sidebands, or partials in an FM potential spectrum are determined by the C:M (frequency) ratio. See FM C:M ratio.

FM sidebands • Partials produced due to linear frequency modulation of an audio frequency periodic carrier signal by an audio frequency periodic modulator signal.

FM C:M ratio • The ratio of carrier and modulator frequencies in an FM operator pair. If the carrier is tuned to 1000 Hz, and its modulator is tuned to 4000 Hz, then the C to M ratio (C:M) is 1:4 (one to four). • Most popular implementations provide a "fixed ratio" mode that keeps the selected C:M ratio the same regardless of which note is played on a controlling keyboard, which maintains the same potential spectrum, and therefore a consistent tone color across the keyboard.

foldover • Alternative term for aliasing. See aliasing.

foley • Sound recordings of real time performances by specialized foley artists who create sounds in synchrony with actors' movements in motion pictures (film, video, etc). The process requires screening of the visual portion of a film, videotape, or digital video silently (without dialogue or SFX) during post production, or after filming is completed. Foley artist(s) are recorded as they create credible descriptive sounds in synchrony that reflect the actions that actors execute on screen. Foley is often intended to represent sounds (e.g. footsteps, punches in fights) caused by human actions portrayed on/off screen. Named after the pioneering foley artist Jack Foley. Cf. SFX. See post production. See foley sound stage.

foley sound stage • A specialized recording studio where foley for film is performed by foley artists in synchrony to picture. A foley sound stage typically has various "pits" filled with gravel, sand, water (!), where foley artists can walk (or splash), and otherwise use various noisemaker props and devices to provide the sounds that human beings depicted in a film make through their body movements in conjunction with their shoes, clothing, jewelry, doors, weapons, etc.

force • Force is the amount of energy expended due to a physical action or movement. For instance, MIDI after touch data actually represent the alterable amount of force continuously expended while key(s) are depressed, whether force is exerted over a particular surface area of the actual keys depressed, or not. For instance, force might remain the same value whether a large human finger or a thin pencil were used to depress key(s). Surface area on the keys depressed plays no role in MIDI after touch (as does "pressure.") Therefore, use of the term "pressure" in the MIDI protocol specification to describe after touch is dubious at best. (Pressure involves surface area). But it is equally doubtful that the more accurate term force will be adopted, even should the MIDI protocol be updated. See after touch. Cf. pressure. See qwerty principle.

force sensitivity • See after touch.

formant • Fixed frequency band(s) in a musical instrument or human voice that emphasize (boost) any partial(s), whether harmonic or inharmonic, that exist in those band(s), regardless of which fundamental (or lowest) frequency (note) is played or sung. • The human voice produces vowels (e, o, u, etc.) because the requisite 3–5 formants for each vowel can be produced by malleable vocal tract anatomy. A complex waveform, e.g. narrow pulse wave mimics the sound produced by human vocal chords. Various boosted filter bands with appropriately tuned center frequencies mimic formants (emphasized frequency bands) as shaped by the vocal tract. The human voice glides smoothly between different sets of formants in order to make transitions between various vowels. • For instruments, formant (band pass center) frequenc(ies) depend on fixed physical dimensions and other typically unchanging aspects of the instrument's size or length, resonant cavity or bore characteristics, as well as overall shape. Therefore, instrumental formant frequenc(ies) are truly fixed. Instrument formants provide important cues that facilitate distinguishing among orchestral instruments. The oboe has a formant centered at 1 kHz; the bassoon has a formant at approximately 500 Hz. • Electronic modules such as band pass filters, graphic EQ, or parametric equalizer (EQ) can be tuned to boost selected frequency bands to create formants in a complex waveform that imitate acoustic instruments, vowels of human speech, or create new sounds. See filter, band pass. See EQ, parametric. See EQ, graphic. See vocoder, channel.

formant frequencies, vocal

fourier analysis • Means of mathematically deconstructing a complex periodic waveform from the time domain, into a collection of constituent sine waves with amplitudes, frequencies, and phases that can be depicted in the frequency domain. The mathematical basis of the principle was stated in 1822 by Jean Baptiste Joseph Fourier (1768–1830 pronounced for´yay) and brought to practical fruition for sound analysis by Georg Philip Ohm (1787–1854), of Ohm's Law fame, and the German physicist Hermann Ludwig Ferdinand von Helmholtz (1821–1894). See FFT.

fps • Acronym (pronounced fips) for frames per second (fps), an indication of frame rate for video or film media, e.g. 25 fps for film in EU and 24 fps in USA. Also appears capitalized (FPS).

free field • With respect to sound, a space or location that does not reflect sound waves to a great extent, e.g. a salt flat or dry lake bed. See inverse square law.

frequency • Number of waveform cycles or physical vibrations of a periodic wave that occur in one second. Frequency is expressed by the unit of measurement Hz, an abbreviation for hertz (after Heinrich Hertz, an early radio experimenter). See (measurement unit) hertz.

frequency response • A measurement that describes the electrical passband of a processor (filter, amplifier, etc.), i.e. how it amplifies or attenuates all possible sine waves within its passband. These data are typically presented as a "frequency response curve" that illustrates what is more accurately an amplitude as a function of frequency curve. To ascertain such a curve, the DUT (device under test) is "swept" by inputting a constant amplitude sine wave that progresses continuously in frequency from, e.g. 20 Hz to 20kHz. The resulting so-called "frequency response" of the DUT is a composite of an individual frequency-by-frequency comparison of the amplitudes of input versus output sine waves. A device with a "flat" frequency response, e.g. and "ideal" amplifier, is graphed as a horizontal line stretching from lowest to highest allowable frequency, indicating that all frequencies input to such a device are amplified (or attenuated) by the same factor. A typical filter has a frequency response curve that shows selective attenuation (or amplification) of specific frequency bands, and usually produces a curve.

frequency, alias • See alias.

frequency, boundary • Translation into hertz (Hz) of the equivalent wavelength, or lambda (□) analogous to the size (diameter) of an opening, or aperture through which partials of a complex waveform pass. Partials with shorter wavelengths, i.e. frequencies higher than the boundary frequency "beam," or move essentially straight through the aperture. Partials with longer wavelengths, i.e. frequencies lower than the boundary frequency "bend," or diffract as they pass through the aperture. See (wave propagation) diffraction. • The formula that expresses the relationship of the velocity (v) of sound (344 m/s @ 20° C), frequency (f), and wavelength (□) is: v = f □ (spoken "velocity equals frequency times lambda," recalled by the memory aid "very fine liquor" where "l" stands for lambda). The bell of a brass instrument is an aperture with a diameter whose size can be equated to an equivalent wavelength, or lambda (□). The equivalent boundary frequency for a bell with a diameter of, e.g. 0.40 meters (40 centimeters, or about 16 inches) is determined by the previous formula, rearranged as: f = v / □. In this example f = 344 / 0 .40 which yields a (boundary) frequency of 860 Hz. Partials with frequencies higher than this boundary frequency (860 Hz) tend to beam, or go straight through the bell. Partials with frequencies lower than this boundary frequency (860 Hz) diffract, or bend and spread through an opening of this size. • The boundary frequency is a useful concept, but it does not necessarily sharply differentiate beaming from diffracting. That is, some diffracting and some beaming of partials close to the boundary frequency take place. But, in principle, diffraction occurs when the wavelength of a partial is longer (larger) than the size of the aperture through which it passes. The beaming vs. diffracting boundary is expressed as a frequency equivalent to an aperture size in order to facilitate comparisons with partials, typically harmonics produced by musical instruments, that equate to frequencies. See (wave propagation) diffraction.

frequency, center

frequency, cutoff • Cutoff frequency (f_c , spoken “f sub c”) is the unique frequency on a cutoff filter slope that marks the boundary between pass band and stop band. The cutoff frequency’s amplitude is 3dB lower than those frequencies not attenuated, i.e. frequencies lying well within the pass band. Because a 3 dB loss (–3dB) is a halving (½) of power, the cutoff frequency is aka the half power point. A partial precisely at the half power point has an amplitude that is approximately 71% (0.707) of the amplitude of those partials in the pass band that are not attenuated at all. Halving (½) of power is 0.5 in decimal form. But where does “0.707” come from? It derives from the square relationship between amplitude (0.707) and power (0.5), expressed ($0.707^2 = (0.707)(0.707) \approx 0.5$). Conversely, the square root of power yields amplitude ($\sqrt{0.5} \approx 0.707$). See relationship, square. See filter slope. See pass band. See stop band. See half power point.

frequency, negative • Frequency with a signal polarity (–) opposite that of a frequency deemed or determined to be positive (+).

frequency, Nyquist • The frequency that is ½ (one half) the sample rate (SR), or sampling frequency, aka the Nyquist limit, or simply the Nyquist. Any partial whose frequency exceeds the Nyquist will not be sampled accurately, but will produce an alias, or partial with a spurious, or incorrect frequency. See alias. See aliasing. The period (T), inverse of frequency (f), of any partial whose frequency exceeds the Nyquist will consequently have fewer than two samples, because Nyquist = SR / 2 where SR is the sample rate. The audio compact disc (CD) debuted with a sample rate of 44 100 Hz, with a resulting Nyquist of 22 050 Hz.

frequency band • Group of contiguous, or adjacent frequencies, whose bandwidth is defined by the lowest and highest frequencies at the extremities of the band. See bandwidth.

frequency counter • Test equipment that displays the frequency of the signal at its input in hertz (Hz).

frequency domain • See domain, frequency. Cf. domain, time.

frequency follower • Processor, circuit, or algorithm that determines the frequency of an input (typically AC) signal, and outputs a stepped or relatively slowly varying signal that represents frequency changes of the input signal. For example, a pitch-to-MIDI converter derives MIDI note numbers from the fundamental frequency of an input audio signal. These note numbers may be used to control synthesis modules to make them "track," i.e. respond to the frequency follower (FF) output signal that is proportional to the frequency of the input audio signal. For instance, an oscillator might be made to proportionately follow the frequenc(ies) produced by a human voice using a frequency follower.

frequency modulation • Change or deviation of frequency, particularly of a carrier signal in a communications or sound synthesis system. See synthesis, frequency modulation.

frequency range • A specific band, or span of contiguous, or adjacent frequencies. For instance, AM radio in the USA has been assigned carrier frequencies from 535 000 to 1 605 000 hertz, i.e. from 535 – 1605 kilohertz (kHz). • Audio engineers refer to loosely defined low, mid, and high frequency bands within the audio band of frequencies from 20 Hz – 20 kHz. This audio frequency band lies within a specific, narrow range of the total frequency band of the electromagnetic spectrum, which includes X rays, gamma rays, light, radio waves, etc.

frequency response • The amplitude per frequency response of a device that generates, transmits, transduces, or processes partials or bands of partials within its designated frequency limits. A frequency response curve is a graphic representation of a device's frequency response that should correspond to published design specifications that indicate whether a processor or transducer alters the amplitudes of partials uniformly, e.g. an ideal amplifier, or discriminates among partials by attenuating or boosting selected bands of frequencies, e.g. a cutoff filter. • A specification for audio equipment illustrated graphically in the frequency domain, or listed by largest (±) deviations in dB from a horizontal straight line that illustrates a theoretical perfectly flat frequency response. For instance, a specification of "±2.5 dB departure from flatness" means that the frequency response of that device diverges by no more than 2.5 decibels above (+) or below (–) a horizontal line that represents perfectly flat frequency response. See frequency response, flat. Most graphical depictions show power or amplitude in decibels (dB) on the vertical axis, and logarithmic (log) frequency on the horizontal axis. Frequency response curves are particularly useful for graphically characterizing various filter types. See frequency response, flat. See function, logarithmic. See frequency representation, log.

frequency response curve • Graphical representation of the frequency response of a device. A frequency response curve is typically shown on semi-log graph paper with log frequency on the horizontal (x) axis, and decibels on the vertical (y) axis. See frequency representation, log. Although the expression "frequency" response curve is commonly used, a more descriptive expression would be “amplitude versus frequency” response curve. See frequency response.

frequency response, flat • Frequency response of a device that treats the amplitudes of all partials generated, transduced, transmitted, or processed equitably, i.e. uniformly. Such a “flat” response appears as a horizontal line on a semi-log depiction of a frequency response curve. For a generator, e.g. noise source, flat frequency response means that all output partials or bands of partials within design limitations have equal power, represented either by log frequency (pink noise) or linear frequency (white noise) using a frequency response curve. Only ideal devices achieve a perfectly flat frequency response; real world devices diverge from perfect flatness. See noise, pink. See noise, white. See frequency response curve. See frequency response.

frequency representation, linear • A graphic depiction of frequency on the “x” axis such that any numerically equal frequency band occupies equal space graphically, is a linear representation. For example, frequency bands at 1kHz-2kHz and 10kHz-11kHz occupy equal space (distance graphically) in a linear representation. On the other hand, any constant frequency ratio (e.g. a musical interval) occupies progressively larger amounts of space in a linear frequency representation as frequency increases. For instance, the octave from 2kHz-4kHz occupies twice the space as does the octave from 1kHz-2kHz in a linear representation of frequency. An ordinary ruler or “yardstick” depicts adjacent numerical markings in terms of the same spatial unit (e.g. inches) and is therefore a linear representation of distance. • A piano keyboard is a linear representation of musical interval. For instance, an octave occupies the similar spatial units on the keyboard in any frequency range. However, the piano keyboard is a log, or logarithmic representation of the frequencies that successive octaves encompass. Cf. frequency representation, log.

frequency representation, log • A graphic depiction of frequency on the “x” axis, such that any frequency ratio (e.g. musical interval) occupies an equal amount of space, is a logarithmic (log) representation. • A log representation is a constant percentage display, since a ratio can be expressed as a percentage (%). For instance, an octave (downward) is a 1:2 ratio, which translates to the fraction ½ (one half), whose decimal form 0.5 reveals the percentage 50% (fifty percent). Successively lower octaves comprise 50% reductions of frequency. Constant percentage changes (e.g. same musical interval) that are represented by equal spatial units constitutes a log representation. • A piano keyboard is a logarithmic representation of frequency, and a linear representation of musical interval. Cf. frequency representation, linear.

frequency representation, semi-log • A standard frequency response curve, which might more accurately be called an “amplitude versus frequency” response curve, is typically depicted using a logarithmic frequency representation on the “x” (horizontal) axis. On the “y” (vertical) axis, amplitude, e.g. in voltage or decibels would be represented linearly, hence the term “semi-log” for this type of graph, with one linear axis, and the other logarithmic. See frequency representation, linear. See frequency representation, log.

frequency, resonant • See resonance.

frequency, sampling • Rate at which samples are taken when digitizing an analog signal, to produce a digital file that can be stored in computer memory or digital storage media. • Sampling frequency is expressed in hertz (Hz) and is aka sample rate (SR). • The audio compact disc (CD) debuted with a sample rate of 44 100 Hz. Another early standard for audio SR is 48 000 Hz. Higher sample rate(s), e.g. 96 000 Hz or 192 000 Hz, allow sampling of partials using a proportionately higher Nyquist frequency, which reduces the probability of aliasing (a type of signal distortion when sampling.) See frequency, Nyquist. See aliasing. A higher SR also allows use of a final output low pass, or smoothing filter whose slope is less steep, facilitating less severe phase anomalies amongst partials in the final output. See filter, smoothing.

full scale • That level in a system that represents the largest allowable magnitude (number) or level (signal), a number or level that provides no headroom. See headroom. In digital audio and/or virtual systems, full scale is often arbitrarily taken to have a value of one (1.0). Any signal or number than exceeds full scale in a digital system results in overflow (computer-speak) or clipping (audio engineering-speak). In a digital system, any excursion above full scale is likely to create objectionable distortion (clipping). In analog systems, nominal full scale, e.g. “0” on a VU meter, typically provides “head room” for signals larger than this nominal full scale, with attendant distortion that is not only tolerable, but sometimes desired by the audio engineer.

function

function, exponential • Mathematical relationship where a fixed positive number other than one (1), known as the base is raised to a power by an exponent, or logarithm. • A curve of increasing returns results when an exponential function (y) is graphed, depicting a progressively larger increase in magnitude as (x) increases, based on the general form: $y = a^{x}$ where “a” is the base and “x” is a positive logarithm representing a given power. That is, y is a function of x for any base selected. In the expression $10^{2} = 100$ (spoken “ten squared equals one hundred”), “10” is the base (a), “2” is the power (x), and “y” is a function of “x,” that is: ($y = a^{x} = 10^{2} = 100$). A curve of decreasing returns results when a “decaying” exponential function (y) is graphed, depicting a progressively larger decrease in magnitude as (x) increases, based on the general form: $y = a^{-x}$ where “a” is the base and “x” is a negative logarithm representing a given power. • The Bel, a unit based on the logarithm of the ratio of two powers, yields ten-fold increases or decreases of power when the logarithm (expressed in Bels) increases as a series of whole numbers, e.g. 0, 1, 2, 3, 4, etc., respectively when this series of logarithms is exclusively positive or negative. The Bel has a base of 10 raised to various logarithms, or exponents, thereby creating an exponential power curve. For instance, one Bel is $10^{1} = 10$, a power ratio of 10:1; two Bels is $10^{2} = 100$, a power ratio of 100:1, three Bels is $10^{3} = 1000:1$, etc. As the exponent shown in superscript “x” representing the number of Bel(s) increases, an exponential function (and curve, when graphed) is illustrated.

function, linear • Mathematical relationship of the general form $y = ax + b$ which yields a straight line when depicted on graph paper that is linear in both axes. So, "y" is a function of "x", with constants "a" and "b." Such constants do not change during a particular scientific experiment or mathematical operation but are free to be given different values for each new operation or experiment undertaken. The term "a" is the slope, or steepness of the line produced. That is, the transfer function of a device with positive gain, e.g. 2:1 amplifier would be expressed as: $y = 2x$ (where "x" is the input signal, and "y" is the output). Conversely, a reduction of 50%, a negative gain, or attenuation such as that provided by a 1:2 amplifier (attenuator) would be expressed as: $y = ½ x$ (again, where "x" is the input, and "y' is the output). A patch cord would be expressed as the function: $y = x$ (where one (1) is understood to be the "a" value). The "b" constant determines the vertical (ordinate) axis intercept on the graph, and nonzero "b" values do not pass through the coordinate pair location (0,0) at the center of a four quadrant Cartesian plane, i.e. a typical two dimension (2D) graph with linear axes. The "b" term could be considered a DC bias, or in virtual terms, a constant that is algebraically added to all numbers on the line produced by the linear function $y = ax$. • The hallmark of a linear function is that no term in its equation is raised higher than the first (^{1}) power. Squared (^{2}), cubed (^{3}), or higher powers, i.e. exponents larger than one (1) are not part of a linear function. • A processor designed to be a linear device (e.g. amplifier, filter), does not output frequencies that are not present at its signal input, at least under ordinary conditions. However, a nominally linear device may distort its input signal when the level of that signal exceeds design specifications. In this case, a nominally linear device such as an amplifier can momentarily act like a nonlinear device, outputting partials that are not found in the original input signal. See distortion, nonlinear. See distortion, clipping.

function, logarithmic • Mathematical relationship where a logarithm, or exponent "x" is the power to which a given base (e.g. 10) that can be any positive real number other than one (1), must be raised to express a given number "y," the antilog. • A curve of decreasing returns of the general form, for common logarithms is: $y = \log_{10} x$. When the base is 10, as indicated by the subscript ($_{10}$), common logarithms are in use. In the familiar equality $10^2 = 100$, "10" is the base and "2" is the power, or logarithm to which 10 must be raised to yield "100," the antilog. In logarithmic form of this is shown as: $\log_{10} 100 = 2$ (spoken "the logarithm of one hundred to the base ten equals two"). That is, 10 must be raised to the second (^{2}) power to equal 100. • A logarithmic, or log curve is produced by the attack (A) segment of an analog ADSR envelope generator, historically an indication of how a capacitor that is part of ADSR circuitry is charged. Some virtual designs provide selection among log, linear, or exponential curves for particular envelope generator (EG) segments. Cf. function, exponential. See function, linear.

function, nonlinear • Mathematical relationship that yields a curve rather than a straight line when graphed. Cf. function, linear. • A nonlinear processor (ring modulator, waveshaper, etc.) is designed to produce frequencies at its signal output that are not present at its signal input. Some nonlinear devices, e.g. ring modulator produce clangorous sounds that include inharmonics, derived from nonlinear processing of signals that may contain only harmonics. See clangorous. See (partial) inharmonic. Cf. (partial) harmonic.

function, transfer

function generator • Generator that produces periodic waveforms, e.g. sawtooth, pulse, triangle, sine, based on geometric relationships of sine and/or cosine trigonometric functions. • An "oscillator," although this term was first used to refer to a periodic generator that produced only the sine waveform.

fundamental • The first harmonic (H1) in a harmonic series or a group of harmonics in a complex waveform. The fundamental is aka the "ground tone," but this term is outmoded, as is the term "overtone," meant to describe a harmonic higher than the fundamental. A harmonic series based on a given frequency, has a fundamental (H1) that is perceived as the pitch. This is the case because any two adjacent harmonics lie one fundamental frequency apart from each other, which reinforces the fundamental (H1) frequency as the pitch perceived. That is, any two adjacent harmonics in a harmonic series based on a fundamental of 100 Hz, have a frequency difference of 100 Hz. The fundamental (H1) is the pitch perceived, but not because it may be the partial with the largest amplitude (as is the case with several geometric waveforms). That is, human perception of pitch does not require the physical presence of the fundamental (H1) to establish pitch. See residue pitch. • A bell sound may contain both harmonics and inharmonics. What may "mathematically" appear to be the fundamental frequency does not necessarily dictate the pitch we actually perceive, particularly in a "klang" tone such as a bell. The partial with the lowest frequency in a "klang" waveform, or even the lowest harmonic does not necessarily dictate the pitch we perceive. Pitch perception is complicated, but a clear sense of pitch depends largely on the presence of sufficient numbers of harmonics, or partials that are very nearly harmonic, that have sufficient strength (amplitude) to reinforce the frequency of the fundamental, or first harmonic (H1) as the pitch perceived. Most "tuneful" (orchestral) musical instruments produce a complex tone laden with harmonics, which accounts for the clear sense of pitch they convey. See (partial) harmonic. See (partial) inharmonic.

gain • The increase or decrease of a signal's amplitude, level, or power, represented by the ratio of a processor's signal output (O) level or power compared to its signal input (I) level or power (O:I ratio). See gain stage.

gain stage • One of several points in a signal flow path, e.g. on a recording console where a module, section, device, or electronic component provides positive, negative, or unity gain. See gain. A pad, or switchable fixed attenuator is a negative gain stage when engaged. An audio input amplifier typically is a positive gain stage. See amplifier, positive gain. See amplifier, negative gain. See amplifier, unity gain.

gate, logic • A logic gate is a processor with one or more input(s) that function as threshold detector(s), and a single output that provides only one of two possible step signals conditionally, i.e. based on the “high-low” (level) status of the signal input(s). See logic gate, etc. See Boolean operations. See threshold. See threshold detector.

gate • A gate is an electronic switch that allows a signal to pass through a device, or current to flow in a circuit. The term “gate” is also used by some to indicate an amplifier, particularly a voltage controlled amplifier (VCA), i.e. a broadband processor whose signal output level may be determined by continuous rather than discrete control signal(s). The authors tend to avoid this usage of “gate” to describe any amplifier module, preferring multiplier, signal controlled amplifier (SCA), or voltage controlled amplifier (VCA) terminology.

gate input • An input that embodies a threshold detector that responds to any input signal in a discrete, binary way. When the input signal exceeds the level of the threshold (number, voltage, etc.) input produces the “high” or binary “1” signal or condition. When the input signal does not exceed the threshold level, this type of input produces the “low” or binary “0” signal or condition. Examples include the gate input on an ADSR envelope generator (EG), which functions as a timing (T) input. See ACT. See ADSR. • Note that a gate input is not limited to connection of a gate signal as such. See gate signal. That is, a gate input accepts any signal in the appropriate amplitude range, including continuous signals, e.g. output of a microphone (mic). It is the response of a gate input that is discrete, i.e. binary, but this input response does not dictate that the input signal must have a binary step signal, or discrete signal characteristic. Although signal characteristics of a gate signal as such conform to the discrete input response of a gate input, continuous signal(s) as well as discrete signals may be connected to a gate input usefully. See gate signal. Cf. trigger input.

gate signal • A gate signal is an aperiodic, bi-state step signal that remains at one of its states until changed to the other. In contradistinction, a trigger signal briefly remains at one of its states prior to automatically returning to its other state. See trigger signal. A gate signal is a signal step, or level that is latched until its alternative state, or level is initiated or caused. An obvious example of a gate signal is the note on/off signal produced by a keyboard. A “note on” (MIDI or otherwise) is a gate signal that is “latched” on as long as a key is depressed. A “note off” (MIDI or otherwise) is a gate signal that is “latched” off when that key is released. Cf. trigger signal.

generator • See (module) generator.

generator, 1/f • A type of random signal generator. See noise, 1/f.

generator, aperiodic • Class of modules that output aperiodic signal(s), i.e. those that have no specific period, or consecutive interval(s) of time during which the generated waveform repeats indefinitely. • Aperiodic generators include modules that output random signals, e.g. white noise, pink noise, 1/f noise; and those that output nonrandom signals, e.g. envelope generator. That is, all random signals are aperiodic, but not all aperiodic signals are random. See generator, envelope. See ADSR. Cf. generator, periodic.

generator, envelope • Module that outputs a nonrandom, aperiodic signal when its function is initiated, i.e. gated or triggered by a signal connected to its gate or trigger input. The output signal of an envelope generator (EG) features timed segments, e.g. attack (A), decay (D), release (R), each of which traverses two defined levels in one time programmed by the user. One or both levels of an EG segment may either be programmed by the user, or dictated by the designer. See EG segment. • The EG output signal is aperiodic, i.e. the waveform produced does not repeat automatically. Therefore, an envelope generator (EG) must be signaled when to generate its signal. The EG gate or trigger input accepts timing signal(s) that exceed the gate/trigger input's pre-selected threshold to start the first timed segment (Attack). When the gate (timing) signal falls below the threshold, this starts the last EG segment (Release). See ADSR. See threshold detector. See input response, discrete.

generator, function • Module that outputs periodic geometric waveforms, e.g. sawtooth, pulse, triangle, and sine waveforms. • Analog function generators typically output all waveforms simultaneously. • Digital function generators typically use wave-table lookup technology and have separate lookup tables for each waveform produced.

generator, noise • Module that outputs broadband random signal(s) whose partials exhibit random (aperiodic) phase and amplitude fluctuations. Although partials fluctuate individually and independently, the average amplitude of partials in a (white) noise signal are equal over time. See noise, white. That is, the amplitudes of individual partials at any instant are dissimilar, but averaged over time, all partials have the amplitude indicated by the distribution characteristics, or spectral density of the particular kind of noise generated. Noise generators typically produce white noise and pink noise. Some noise generators can produce various spectral densities that remain random but are neither white nor pink noise, which is achieved by filtering white noise. See noise, pink. See noise, white. See noise, 1/f.

generator, periodic • Class of modules that output periodic signal(s), i.e. those that have a specific period of time during which the generated waveform repeats indefinitely.

generator, pseudorandom noise • Virtual module or algorithm that uses a mathematical process to produce a sequence of numbers that apparently fulfills one or more statistical requirements for randomness. However, such a sequence of binary numbers, having typically been calculated by a specific, replicable algorithm, must eventually repeat its sequence, and technically is not a truly random signal. In mathematics, a random signal must be infinite in time, and is therefore essentially impossible in the real world. • For purposes of sound synthesis, modern digital pseudorandom noise generator(s) produce signals that adequately imitate "truly" random signals such as white noise and pink noise generated by analog circuits. Technically, these analog signals may not "truly" be random either. Maybe, if our noise signals are not truly random, perhaps they are at least a bit erratic–like most musicians. Good enough. See generator, noise.

generator, step • Type of module that produces one, or a series (sequence) of user-variable fixed levels, or in virtual terms, constants. These fixed steps are typically used to control various parameters in a sound synthesis system and/or to initiate (timing) actions. • A bias, or constant module produces a single step, or level that may be varied. An analog sequencer, based on ring counter technology, is a module that can produce steps that may be output serially (one after the other). In most designs, advancement through this series requires an external or internal clock signal. See sequencer, step. A keyboard is a step generator that produces steps, such as MIDI note numbers, based on a performer's playing of specific key(s). A sample and hold (S&H) module outputs step(s) derived from sampling the level of an input signal. See sample and hold. See bias, DC. See constant.

geometric waveform • Waveform comprising partials based on simple geometric identities, e.g. sine and cosine functions. See waveform, geometric etc.

glide • Parameter typical of analog synthesizers, featuring continuous, smooth (pitch) transitions between notes played melodically, rather than normal discrete (pitch) steps. The rate of glide from key to key is programmable, and is usually represented by a simple attenuator. Cf. glissando. See portamento. • Glide is a function of processing a keyboard signal with a low pass filter, or integrator–not a feature provided by an audio oscillator per se. Therefore, keyboard glide might control some parameter other than audio oscillator frequency (pitch), e.g. filter cutoff frequency. The normally discrete steps of the keyboard can still glide between keys, and smooth changes of cutoff frequency can be played on keyboard in this example. (This becomes more apparent when the keyboard is not controlling the oscillator(s) being heard. Turn off "keyboard tracking" in the oscillators.) Gliding pitch is a cliché dating from the early days of analog subtractive voltage controlled synthesizers, and this all-too-common keyboard control of oscillator frequency threatens to blunt consideration of alternative options for using this gliding keyboard signal. A keyboard with glide enabled and oscillator frequency have no necessary or compelling relationship, even though the majority of hard-wired designs produce only gliding frequency (pitch) when glide is enabled. So, it is not necessarily oscillator pitch that describes glide–it is actually the keyboard that glides. Connect its CV Output (analog) or Note Number (digital) output where you will. See ACT. Avoid brainwashing. Unless, of course, you have an "Abby Normal" brain that needs to be washed. In this case, see Young Frankenstein (1974), Directed by Mel Brooks, for a cogent discussion of an "Abby Normal" brain.

glissando • Pitch transition(s), particularly on a keyboard instrument or harp, performed from one note to another by playing some or all of the discrete intervening notes with a strumming gesture. Cf. glide. Glissando is a rapid playing of discrete pitches between a starting and ending pitch, typically done by strumming a keyboard with the nail of a thumb or finger. • Glissando is sometimes confused with portamento, characterized by continuous, rather than discrete changes of pitch between two notes sung or played melodically (one after the other). Cf. portamento.

grain •

GUI • Acronym for a graphical user interface (GUI, pronounced goo´ee). A GUI is the set of pictures or images on a screen, panel, video display terminal (VDT), or other graphic display(s) of a virtual (computer) system that allow the user to access, manipulate, store, and recall system parameters. • In hardware, the front panel of an instrument, module, or device functions as a GUI.

half power point • Frequency whose power is reduced by 3 decibels (dB) with respect to frequencies at some defined full scale (FS) level. • The unique frequency on a filter slope whose level is –3 dB relative to frequencies in the pass band that are not attenuated. A 3 decibel (dB) change is a 2:1 or 1:2 power ratio (1:2 is a halving of power). The half power point, aka the cutoff frequency on a cutoff filter slope marks the boundary between pass band(s) and stop band(s). • Half power point(s) are useful in contexts other than filters. Half power points at low and high frequency extremes might be used to delineate the useful bandwidth, or relatively flat part of the frequency response curve of an amplifier. Any amplifier attenuates partials at extreme frequencies, because no design provides infinite bandwidth. In this case, a half power point indicates significant divergence, defined as a 3 dB loss relative to a flat frequency response curve. This may be called a “corner” frequency in this context, rather that a cutoff frequency. See frequency, cutoff. See filter slope. Cf. frequency response, flat.

half wave • Description of the waveform that results from processing a bipolar (±) waveform with a diode, an electronic component that permits current to flow only in one direction. The half wave can be either unipolar positive (+) or unipolar negative (–), but in either case, a portion of the period of the processed bipolar (±) waveform will maintain a level of zero (0) due to the diode’s action. See rectifier.

haptic • Relating to the human sense of touch, and the manipulation of objects using senses of touch and proprioception (relating to feelings or stimuli produced within the body, particular those having to do with orientation and/or location of body parts in space.) • The term has been used to describe the forms of “interfaces” between musician and instrument, and the (rather slow) data rates possible with such interactions compared to the audio frequencies produced. Because human touch involves a response with relatively slow speed, “haptic” is taken by some to describe slower-moving “control” signals in the man-machine instrumental music paradigm, or indeed in electronic systems.

hard-wired • Description of an electronic instrument, system, or group of modules or circuits connected in a permanently fixed configuration, making patching changes difficult or impracticable. • Hard-wired instruments (e.g. some synthesizers and samplers) may allow programming of module parameters, but provide limited freedom to change the patch, i.e. the possible interconnections of available internal modules. • Many of the early analog, hardware non-modular portable synthesizers are programmable, but hard-wired, e.g. Moog Minimoog.

harmonic • A partial (sine or cosine wave) whose frequency is a whole number multiple of the fundamental frequency of a complex waveform. The fundamental is aka the first harmonic (H1). See harmonic series. A partial that is not an inharmonic. A partial (sine wave) that stands at integral (whole number) frequency ratio(s) with other partial(s) or a set of partials such as a harmonic series. See harmonic series. See (partial) harmonic. Cf. (partial) inharmonic.

harmonic series • In musical terms, a set of frequencies (1x, 2x, 3x, 4x, etc.) that are whole number multiples of a given frequency (x). This frequency (x) is aka the first harmonic (H1), or fundamental. For instance, the harmonic series on a fundamental frequency of 90 Hz (1x), has a second harmonic (H2) at 180 Hz (2x), a third harmonic (H3) at 270 Hz (3x), and so forth, i.e. harmonics occur at successive whole number (integer) multiples of the fundamental frequency (H1). • A musical instrument that has a clearly identifiable pitch typically has a complex tone comprising various harmonics, each having a different steady state amplitude (strength). See steady state. Each harmonic is aka a partial, meaning a part or component of the complex tone. The complex tone of a pitched musical instrument does not necessarily contain all possible harmonics. For instance, the clarinet is characterized by stronger (higher amplitude) odd numbered harmonics (1x, 3x, 5x, etc.), particularly in its lowest pitch register. In addition, the amplitude of any partial present in the waveform of most musical instruments is typically dynamic, unlike the waveforms produced by a function generator (oscillator). A square waveform (50% duty cycle pulse waveform) has all odd numbered harmonics, and no even harmonics, and is therefore a good basis for synthesis of a rudimentary “clarinet” (stopped cylinder) sound. See (partial) harmonic. Cf. (partial) inharmonic. See spectrum, line.

Haas effect • See precedence effect.

headroom • In audio engineering, headroom is the difference between the nominally largest allowable input signal amplitude (typically zero VU), and the very largest input level the system will accommodate without producing objectionable distortion. In the case of analog equipment and signals, the plus (+3) dB “red” area above 0 dB on a volume unit (VU) meter is deemed headroom. Distortion that occurs when an analog signal falls in this range may not be entirely disagreeable, due to the nature and onset characteristics of analog equipment distortion. In fact, such distortion may produce harmonics that are deemed to enhance the sound. However, in digital equipment, a signal that exceeds full scale, the nominally largest allowable number, will typically create highly objectionable distortion, and “headroom” in such a context must be designed into the system to fall below that full scale level. See amplitude, full scale. Unfortunately, software designers may “build in” head room into their systems without specifying what those headroom limits or spans are. • Headroom might be thought of as a distance or space, by analogy to the space between the top of a vehicle and the overhead of a tunnel through which that vehicle passes, or the bottom of a bridge under which that vehicle passes.

hum • A low level signal caused by unwanted intrusion of local alternating current (AC) power frequency (nominally 60 Hz USA or 50 Hz EU) and possibly higher harmonics of such frequencies, into an audio production. Hum is audio interference introduced from a source outside the system, in contradistinction to audio distortion, an anomaly already present within, or caused by internal system operating conditions. See (interference) etc. See distortion.

Hz • Abbreviation for the unit of measurement of frequency in hertz (abbreviated Hz with no period), which supersedes the older, albeit descriptive term cycles per second (cps). Frequency in hertz (Hz) is a count of occurrences of something, e.g. waveform periods in a unit of time, i.e. the number of times a periodic waveform completes its 360° cycle, or period (T) in one second. See (measurement unit) hertz. • This unit of measurement is named in honor of Heinrich Rudolph Hertz (1857–1894), a renowned German physicist who was the first to observe and describe Hertzian, or radio frequency electromagnetic waves.

IC • An integrated circuit (IC) is a composite electronic device, or "chip" that includes the circuitry or functions of many discrete components, e.g. resistors, operational amplifiers, capacitors, etc.

icon • In sound design systems, an icon is typically a line drawing that represents a particular module. Icons are used in a block diagram to represent particular connections or a configuration of modules that comprise a "patch." See patch.

i.e. • Abbreviation for id est, Latin for that is.

impedance • A quantification in ohms, the unit of measurement for electrical resistance, of electronic circuit components and other elements that oppose (resist) the flow of electricity in a circuit. Impedance is associated in particular with the measure of resistance to the flow of alternating current (AC). Impedance is represented by the letter "Z," and comprises both DC resistance and AC inductive and capacitive reactances, other forms of resistance. • Strictly speaking, it is a misnomer to refer to cables used in audio engineering as high impedance or low impedance cables. No cable per se used to transmit signals used in audio engineering is purposely designed to resist, or impede the flow of such tiny electrical signals, thereby dissipating signal strength either as heat, and/or through dynamically changing magnetic flux and/or electrical fields. To provide perspective, the heating elements in an electric stove are specifically designed to impede the flow of electricity–to the point of glowing red hot! We don't have cables designed to strongly resist the flow of signals in audio engineering. Rather, an audio cable is equipped with plugs (e.g. XLR) appropriate for connection to high impedance circuitry, or plugs (e.g. RCA phono) appropriate for connection to low impedance circuitry, as per industry conventions. The "high Z" vs. "low Z" distinction made between audio cables derives from the impedance of the devices to which a cable is connected, exemplified by the kind of plugs and jacks provided, not due to any electrical characteristic (e.g. impedance) of the cable itself. • The misconstrued idea that a cable for audio engineering might be designed with high impedance is also, unfortunately, intertwined in some people's thinking with the choice between balanced vs. unbalanced cables as well. See cable, balanced. See cable, unbalanced.

impulse response •

index of modulation • Depth (amount) of modulation in a modulator-carrier system that accounts for the potential production and amplitude(s) of sidebands, i.e. partials (sine waves), particularly as related to rapid linear frequency modulation (FM), and amplitude modulation (AM) sound synthesis techniques.

infrasonic • Term for frequenc(ies) below the lower limit of human hearing (nominally 20 Hz). Not to be confused with the term "subsonic," whose use is inappropriate in the sonic arts. See subsonic.

inharmonic • A partial whose frequency ratio with another partial is not harmonic, i.e. not a rational, or whole number frequency ratio. The term inharmonic is also seen as "nonharmonic." See (partial) inharmonic. Cf. (partial) harmonic. • The term inharmonic should not be confused with enharmonic, which relates to the way notes in a musical chord or tonal key are notated, i.e. "spelled." See enharmonic.

input • Any port, jack, or "female" device to which a cable, patch cord, or other male device may be connected to transmit signal(s) into a module or electronic circuitry having such an input. • A point at which data may enter a system or module.

input, audio • An input directly connected to a sound producing system, i.e. an audio monitor. An audio input typically accepts a bipolar signal.

input, control • An input that accepts a signal in order to change, alter, i.e. control a particular parameter or action of circuitry or its virtual equivalent within a module. A control input typically responds in a continuous–rather than discrete manner to input signals. See input response, continuous. • Signals connected to a control input of a module do not generally flow out of the signal output of that module.

input, gate • An input that constitutes a threshold detector, that typically initiates some function(s) or action(s) provided by a module. See input response, discrete. Cf. input, timing. See threshold detector.

input, normaled • An input that is "normally" connected to a specific output but allows the possibility of selection of a different output. The normaled input (or output) exists to provide the most commonly needed connection, while providing the flexibility to make a different connection between signal source and destination. See normaled input. cf. normaled output.

input, signal • Signal input identifies the input on a processor that provides a path through that module to its corresponding signal output. Signal input is distinguished from other kinds of inputs (control or timing) on any processor, such as a signal controlled amplifier (SCA), or signal controlled filter (SCF) on this basis of signal flow. Of necessity, any generator or processor has a signal output, but a generator has no signal input. However, a generator may have a control input, e.g. signal controlled oscillator

(SCO), and some generators, e.g. envelope generator (EG) necessarily have a timing input. But, unlike a signal input, a control or timing input does not provide a path to the signal output of a module, whether it is a processor or a generator. Nor does a control or timing input have an associated control or timing output. • The nomenclature (identifying alphanumeric names and symbols) for a "signal" input purposely and necessarily does not specify the input signal's function as audio, control, or timing (ACT). Why? Because any module's "signal" output might be connected variously to audio, control, timing (ACT) inputs. When a connection is actually made from a processor's signal output to a specific ACT input, the processed signal then functions in that specific way. This begs the question as to how the "signal" input of the processor, or indeed even its "signal" output might be named prior to making such connection(s). Is the signal input on a processor an "audio, control" or "timing" input? It depends strictly on where the signal output of the processor is connected at any given moment. "Signal" input is sufficiently general to account for any signal output connection (ACT), and "signal" input nomenclature also provides an important distinction about signal flow. A signal input has a corresponding signal output, indicating flow through a module. A control or timing input on a module has no corresponding control or timing output, and this nomenclature indicates that a signal connected to a control or timing input does not flow through the module. • A processor simply processes the signal connected to its signal input–without reference to where its signal output is connected. For instance, low pass filter (LPF) processing of a stepped keyboard signal (e.g. MIDI keyboard note number) can "smooth" such a signal, creating "glide" (portamento) when note number controls audio oscillator frequency. Such a filter that clearly functions as a control signal processor, should not legitimately have its signal input designated as "audio!" More conventionally, the filter's signal output might be connected to an audio input, but it certainly could be connected to a timing input as well. Properly considered, a filter should not be categorized as an "audio" signal processor. A filter is simply a processor that might filter any kind of signal–audio, control, or timing (ACT). So, prior to connecting a processor's signal output to ACT input(s), it is very likely contradictory to designate that processor's corresponding signal input somehow other than generically as a "signal" input. ["Let's see, the input says 'audio' on this amplifier (VCA), but I'm using it to change the amplitude of a control signal that shapes vibrato depth. What gives?!! I thought that "amplitude" means "loudness." But that's not what I'm hearing! Hey, Tom and Dave did warn me about falling prey to inappropriately applied signal-sound correlates."] Processors should have nomenclature that clearly designates a signal input and a corresponding signal output. The signal input designation is generic rather than specific, because functionally it must be, as we do not necessarily know where that processor's signal output might be connected. A processor or generator may have other inputs (control, timing) as well, but such inputs behave differently in terms of signal flow, than does a signal input, as discussed above. See ACT.

input, timing • An input that responds to input signals in a discrete manner, i.e. as a threshold detector. A timing input typically determines how an event or function in time behaves, particularly when a particular signal element begins. See threshold detector.

input response, continuous • The mode or manner than an input responds to input signals. Most inputs with a continuous response, e.g. oscillator frequency control, filter cutoff frequency control, audio inputs, etc. exhibit a response that is sensitive to and provides an unlimited (continuous) number of conditions or responses to any polarity (±) and magnitude of an appropriately scaled input signal. • Many inputs that feature continuous input response (e.g. SCO frequency control input) are sensitive to bipolar signals, responding to both positive (+) or negative (+) signal levels. Some inputs that feature continuous input response (e.g. two quadrant SCA modulation input) are sensitive to only unipolar signals, typically responding to only positive (+) signal levels.

input response, discrete • The mode or manner that an input responds to input signals. Inputs with a discrete response, e.g. timing inputs, logic gate inputs, etc. exhibit a threshold response, providing a binary, or limited number of conditions or responses to an input signal. See input, timing.

inputs, differential • See jacks, differential input. Cf. jacks, differential output.

integer • Any negative (–) or positive (+) whole number such as –4, –3, –2, –1, 1, 2, 3, 4, etc. "Counting" numbers are integers. An integral relationship is expressed using integers, or numbers that can be made equivalent to whole numbers, e.g. the ratio 0.5:1 can also be expressed as the integral ratio 1:2.

integral • Relating to an integer or integers (whole numbers). • Relating to mathematical integrals (calculus) or integration. See integer.

intensity • Power of a sound or audio signal, as measured in Bels or decibels (dB). • Sound intensity level (SIL) in decibels is ten times the logarithm of the ratio of two selected intensities, or powers: $SIL_{dB} = 10 \log (W_1 / W_0)$ with power in watts (W). See watt. See (measurement unit) decibel. • Intensity is the chief objective correlate to our subjective perception of loudness, when other signal characteristics such as frequency, waveform, and duration are held constant. See correlate.

interference, constructive • Algebraic (±) summation of two or more waves at a point in space (a room), or at an instant in time in a circuit, virtual or physical, that yields a larger level than either of these summed waves exhibits individually at that point or time. Cf. interference, destructive. • See (wave propagation) interference.

interference, destructive • Algebraic (±) summation of two or more waves at a point in space (e.g in a room) or at an instant in time in a circuit, virtual or physical, that yields a smaller level than either of those summed waves exhibits individually at that point or time. Two waves with precisely the same frequency and amplitude, having opposing polarities (±) or having phases 180° apart, would theoretically cancel each other completely. This is possible on a "wire" (electronically), but unlikely in a room, where sound waves interfere both destructively and constructively due to many reflections from room surfaces. However, it is likely that different locations in a room will exhibit lowered amplitudes for a given frequency due to destructive interference. Conversely, constructive interference causes locations in a room for the same frequency for which the amplitude is greater. See interference, constructive. Such destructive and constructive interference plays a large role in the room's resulting frequency response. See frequency response. See room tone. See (wave propagation) interference. Cf. interference, constructive.

(interference) EMI • Electromagnetic interference (EMI) is an unwanted intrusion into an electronic circuit or system by electromagnetic signal(s) or energy due to radiation energy coupling. Alternating current (AC) power lines in close proximity to audio cables can induce such EMI. It is good practice to position power and audio cables in the recording studio at 90 ° (ninety degree, or right angle) relationships where possible to minimize such interference. See (interference) hum.

(interference) hum • Unwanted intrusion into an electronic circuit or system of a low frequency signal (60 Hz USA, 50 Hz EU), and/or harmonics (whole number multiples) of such frequencies, derived from alternating current (AC), or electrical power. Hum is typically caused by inductive coupling of magnetic flux (changes) caused by alternating current (AC) in one conductor (power cable) that induces voltages in another conductor (audio cable and/or circuit board trace). See (interference) hum.

(interference) RFI • Radio frequency interference (RFI) is unwanted intrusion into an electronic circuit or system, by a carrier in the radio frequency range of the electromagnetic spectrum or by the program signal being carried. Such electromagnetic interference (EMI) may be caused by natural sources such as cosmic or solar sources (sunspots), but RFI is typically man-made, e.g. the electromagnetic "smog" caused by myriad transmissions of radio and television signals that typically originate in urban areas. In some cases, the program signal (not the carrier), e.g. the voices carried by local mobile radios (taxi cabs) interferes with audio productions. In this case, some element of the system, e.g. small discontinuities, or breaks in a cable or circuit board trace may function like a diode, which is the basis of a rudimentary radio receiver. Check your cables and connections for minute discontinuities if you are picking up local radio!

interpolation • Insertion of calculated values between known surrounding, or "bracketing" values, with the intention of correcting errors (compact disc (CD) playback); improving signal recovery (oversampling); or smoothing a transition between two digital data streams (digital signal processing, or DSP), thereby facilitating cross fading of signals, and other editing operations. Previous value interpolation simply repeats the previous value when data are missing. First order (linear) interpolation computes the arithmetic mean (average) between two existing samples or values to create the interpolated value. Higher order interpolations use polynomials to compute intervening values.

interval, musical • Relationship of two frequencies that represents the distance between two notes in a particular musical scale, or temperament system. There are many temperament systems, but all feature ratiometric frequency relationships called intervals. See temperament. That is, each frequency of a pair of notes that constitute a particular interval is a multiplication, or factor of its counterpart. See factor. • A musical interval is defined by the ratio of its two frequencies. For example, a frequency ratio of 2:1 or 1:2 represents the musical interval of an octave. That is, multiplying any frequency by a factor of two (2) constitutes a ratio of 2:1, which produces a frequency one octave higher. Conversely, multiplying any frequency times one half (½), a ratio of 1:2 (the reciprocal of 2:1), produces a frequency one octave lower. So, both frequency ratios 2:1 and 1:2 represent an octave, but the order of numbers in the ratio indicates the up or down direction of the interval relative to the original frequency. • The integer (whole number) ratios of any harmonic series produce "just temperament " intervals, e.g. 4:5:6 is a major triad in just temperament.

inverse • The inverse of a given number is one (1) divided by that number. The inverse of a given number x is 1/x. See relationship, inverse.

inverse square law • In acoustics, the description of the change of sound intensity level (SIL) as the distance from a point source of sound such as a loudspeaker in a free (non-reflecting) field changes. The inverse square law indicates a 6 decibel loss (-6 dB), i.e. a quartering of power for each doubling of distance from a point source of sound in a free field. Each doubling (2:1 or 2/1) of distance causes a halving (1:2 or 1/2) of signal amplitude–that's the inverse part of this particular law. Halving (1/2) amplitude is equivalent to quartering (1/4) power, due to the square relationship between signal amplitude and power. That is, one half squared equals one fourth (½ x ½ = ¼), or a quartering of power. A quartering (¼) of power is a loss of six decibels (–6dB). See relationship, inverse. See relationship, square. • The inverse square law is a law in physics for various phenomena other than sound, e.g. relating distance and: force of gravity; electrostatic charges on capacitor plates; lumens of light; retinal image size; etc. There are cube (third power) laws as well, e.g. power of wind as a function of wind speed. A wind speed with three (3) times greater velocity exerts nine (9) times more force (three cubed). Such laws of physics never seem to go out of style, they're not trendy. Might be worth learning about some of them. Again, the wisdom of Yoda. See Star Wars (1977), Directed by George Lucas.

inversion • Process of raising the lower note of a musical interval by an octave, or lowering the upper note of that interval by an octave. See invert. For example, the inversion of a perfect fifth (P5) is a perfect fourth (P4). The inversion of a major interval, e.g. major third (M3) is a minor interval (e.g. minor sixth (m6) in this example) • Moving of the root of a chord (minimum of 3 notes) such that it is no longer the lowest note in the chord. Such a chord is called an inversion of the original chord. • Compositional device that changes the notes in a melody from ascending (up) to descending (down), and the converse, typically maintaining the exact musical intervals involved. Directionality of intervals is changed relative to a particular selected note in the melody (tonic, or "home" note; first or last note, etc.). Sometimes this kind of melodic inversion changes directions within the key involved, and musical intervals may not be strictly maintained. See interval, musical.

invert • The make opposite, to turn upside down. • In music, to rearrange the notes of a musical interval, e.g. M3 (major third), to produce the inversion of that interval, m6 (minor sixth). See inversion.

inverter • Processor that changes the polarity of a signal from positive (+) to negative (–), or the converse. In the case of a bipolar (±) signal, the inverter changes both original polarities to their respective opposite mathematical sign, causing a 180 ° (one hundred eighty degree) change of polarity. • Note that such a polarity change does not constitute a change of "phase." Phase involves a temporal (time) anomaly or condition, such as the delay time between one signal and another. • Electrical device that changes DC to AC, allowing a battery to operate AC devices.

I/O • Abbreviation or acronym for input/output (I/O), (spoken by saying letters "I" and "O"), referring to general access points or terminals for communicating within or between systems or circuits.

ips • Acronym (spoken "ips") for inches per second (ips), representing transport speeds of analog magnetic audio tape recorders, particularly in USA. Popular USA transport speeds include 3.75, 7.5, 15, and 30 inches per second (ips) for analog magnetic audio tape. In European Union (EU), transport speed may be expressed in centimeters per second rather than inches.

jack/plug

jack/plug, banana

jack/plug, BNC

jack/plug, DIN

jack/plug, miniature phone

jack/plug, RCA

jack/plug, phone

jack/plug, XLR

jacks, differential input • Differential inputs are a pair of jacks that feature opposite (±) input signal polarities. One jack inverts, or negates (–) the input signal, while the other is a standard, non-inverting (+) jack. Such inputs may be said to be 180° "out of phase" with each other, but phase is a misnomer in this case. A phase relationship results when one signal is delayed with respect to another, and delay plays no discernible role in the design of differential jacks. Rather, differential input designs invert (negate) signal polarity at one of two input jacks, making opposite (±) plus and minus polarities available at such dual jacks. Cf. jacks, differential output.

jacks, differential output • Differential outputs are a pair of jacks that feature opposite (±) output signal polarities. One jack inverts, or negates (–) the output signal, while the other is a standard, non-inverting (+) jack. Such outputs may be said to be 180° "out of phase" with each other, but phase is a misnomer in this case. A phase relationship occurs when one signal is delayed with respect to another, and delay plays no discernible role in the design of differential jacks. Rather, differential output designs invert (negate) signal polarity at one of two output jacks, making opposite (±) plus and minus polarities available at such dual jacks. Cf. jacks, differential input.

lambda • Eleventh letter of the Greek alphabet (□) used to represent wavelength, a spatial measurement, e.g. in meters (m), e.g. constituting the distance a periodic wave travels during the time of its period (T). • Sound waves of all frequencies (f) propagate (travel) at the same velocity (v) in a given material elastic medium, e.g. air, for which v = 344 m/s @ 20° Celsius. The general formula that relates velocity, frequency, and lambda is: v = f □ recalled by the handy mnemonic "very fine liquor." Frequency (f) and period (T) have an inverse relationship. That is, f = 1/T and T = 1/f. See relationship, inverse. A waveform with a period (T) has a corresponding frequency (f) and wavelength (□). These terms are related by the formula: □ = v / f (spoken "lambda equals velocity (of sound) divided by frequency.") See wavelength. See period. See frequency.

latency • In general terms, the delay, or amount of time between a stimulus and its response. • Amount of delay time in mechanical or electronic systems, particularly the time it takes digital information to propagate (travel) within a computer, based primarily on the temporal overhead (time delays) caused by execution of specific software routines. Because electrical signals as such propagate at a velocity approaching the speed of light, such signal delays play a negligible role in causing latency in the analog part of a computer system. • In a digital sound synthesis system, the time it takes for data to pass through a physical gate, network connection, or (virtual) algorithm causes latency, and such delay(s) can become audible to the point of being obtrusive when dealing with digital audio. Because analog electrical signals propagate at a velocity that approaches the speed of light (186 000 miles per second), properly designed analog sound synthesis systems operate with negligible latency. See delay.

LCD (Liquid Crystal Display)

LED • Light emitting diode (LED), a semiconductor device that produces (typically) colored light when voltage is applied to its input(s). The "power on" light on many hardware telecommunications devices and computer peripherals (e.g. disk drive, DAC, etc.) is typically an LED.

level

LFO • Low frequency oscillator (LFO), a narrowband periodic function generator whose frequency range is designed to produce only lower frequencies, e.g. from infrasonic (below human hearing) through 50–100 Hz. An LFO is an oscillator designed specifically for "non-audio," typically low frequency control or timing functions, e.g. creating vibrato and tremolo, triggering (gating) envelope generators, etc. An LFO may not be signal controlled (its frequency is then set using a constant), but some designs may allow frequency control via external control input(s). However, many nominal audio frequency oscillators might be tuned to their lowest frequency range in order to behave exactly like an LFO. The original distinction between an LFO and a voltage controlled oscillator (VCO) or signal controlled oscillator (SCO) designed to generate periodic audio waveforms was made historically due to hardware cost vs. utility tradeoffs. In a virtual system, this tradeoff is digital signal processing (DSP) overhead vs. utility. That is, an LFO uses less of the total DSP bandwidth available than would a broadband so-called "audio" oscillator, or SCO (signal controlled oscillator). • In the authors' view, the only legitimate distinction that can be made between the function of an LFO and another oscillator with a wider frequency range (e.g. VCO), is determined by the type of input to which either oscillator is connected (see ACT). That is, any LFO can actually function as an "audio" oscillator, albeit with a restricted frequency span. See oscillator. See generator, function.

limiter • Processor that prevents an input signal's level from exceeding a programmable threshold value. Limiting is a severe form of compression. A limiter has a high compression ratio, e.g. 20:1 or greater. That is, at 20:1 every input signal level excursion that is 20 dB above the threshold of a limiter is compressed, or limited to only a 1 dB increase at the compressor signal output. Signal levels below the selected threshold are not affected. That is, signals with levels below the threshold bypass compression or limiting operations. • A compressor or limiter can reduce the dynamic range of the processed signal, mitigating the tradeoff between the twin, opposing evils of distortion vs. noise in audio recording. See compression, audio.

line level

linear •

logarithm • Superscript (raised) number that represents the power to which a given base (e.g. ten) must be raised to express an equivalent number called the antilogarithm, or antilog. Logarithms with base ten (10) are known as common logarithms, and this is the type used in acoustics and audio engineering. Common logarithms express any number, or antilog as a power of ten (10). A common logarithm has two parts, the characteristic whole number (integer) to the left of a decimal point; and the mantissa, or fractional part to the right of a decimal point. For instance, $10^{1.301}$ has a characteristic of "1" and a mantissa of ".301" which produces an antilog of about twenty (20). In this example, the characteristic indicates a 10:1 ratio and the mantissa indicates a 2:1 ratio. Taken together, this constitutes a 20:1 ratio. That is, $10^{1.301} \approx 20$. • In context of the decibel (dB), an antilog such as 20 represents the power ratio (20:1) of interest, the logarithm ($^{1.301}$) represents the equivalent number of Bels (1.301), and equivalent decibels (13.01) for that particular antilog (20), and the base is ten (10). The decibel (dB) is based on common (base 10) logarithms. Any 20:1 power ratio (e.g. 800:40 Watts, 200:10 Watts, 20:1

Watts) is expressed logarithmically as 10 $^{1.301}$ aka 1.301 Bels, approximately equivalent to thirteen decibels (13 dB). See Bel. See (measurement unit) decibel.

logic gate • A processor module or circuit that outputs only one of two possible signal levels, or binary states at any given moment: logical 1 (true) or logical 0 (false). These "levels" are only conceptually "zero" and "one;" i.e. in electronic circuitry such "true" and "false" signals may have electrical levels arbitrarily specified by the designer. Gate output is determined by the condition(s) of signal(s) at one or more associated gate inputs. See logic gate, etc. Electronic signals, or their virtual counterparts typically represent logical 1 (true) using a higher (positive) signal level (e.g. voltage or number) than logical 0 (false) signal. • Logic gate input true (T) or false (F) values change as the threshold that distinguishes between high and low signal states is crossed in either direction. A logic gate input is essentially a threshold detector. As such, it is not sensitive to magnitudes of an input signal other than the two values (0 and 1) immediately adjacent to the input's threshold value, which conceptually falls between 0 and 1. • Logic gate input(s) exhibit a discrete response–high or low. A logic gate output produces a signal having only one of two possible states (0 or 1), but a logic gate input can respond to a signal that has more than two states, i.e. a continuous signal. Because a logic gate input is a threshold detector, it might be fruitfully connected to any signal that varies continuously in magnitude and/or polarity, such as a microphone (mic) output. Audio signals might then be used to "make decisions" in some artistic scenario using gates. The artistic success of such divergent uses of logic gate inputs in music synthesis depends on clever patch design, and intelligent biasing and attenuation of the signal(s) connected to logic gate input(s). See threshold detector.

logic gate, A-AND-NOT-B • A device with two inputs typically labeled A and B, and one output, that produces a logical 1 (true) only when there is a logical 1 (true) at input A, and a logical 0 (false) at input B. That is, the result is true only when A is true and B is false. Other input combinations output a logical 0 (false). • This type of AND NOT gate is also available as a B-AND-NOT-A device that produces a true only when B is true and A is false. Other input combinations output a logical 1 (true).

logic gate, AND • A device with two inputs and one output that produces a logical 1 (true) only when there is a logical 1 (true) at both inputs. Other combinations at A and B inputs produce a logical 0 (false). AND is not an acronym, it literally means "and."

logic gate, A-OR-NOT-B • A device with two inputs typically labeled A and B and one output, that produces a logical 0 (false) only when there is a logical 0 (false) at input A, and a logical 1 (true) at input B. That is, the result is false only when A is false and B is true. Other combinations at A and B inputs produce a logical 1 (true). • This type of gate is also available as a B-OR-NOT-A device, i.e. the output is false only when A is true and B is false. Other combinations of A and B inputs produce a logical 1 (true).

logic gate, NAND • The negation of an AND gate, (spoken as "NAND"), meaning "not and." A device with two inputs and one output that produces a logical 0 (false) only when there is a logical 1 (true) present at both inputs. Other combinations at A and B inputs produce a logical 1 (true).

logic gate, NEGATION • An inverter, or negation circuit with one input and one output. It produces a logical 1 (true) when there is a logical 0 (false) at the input, and a logical 0 (false) when there is a logical 1 (true) at the input. NEGATION is not an acronym, it literally means "negation."

logic gate, NEGATIVE (A-IGNORE-B) • A device with two inputs typically labeled A and B, and one output that produces a logical 1 (true) when there is a logical 0 (false) at the A input, and a logical 0 (false) when there is a logical 1 (true) at the A input. The B input plays no role in determining output. If input B were always functional this would operate as a NOR gate, but in this case the B input can be disabled by a (temporary) signal. Upon receipt of this signal, input B is disabled, and this gate is functions as a NEGATIVE (A-IGNORE-B) gate. When input B is disabled this gate functions like a NEGATION gate.

logic gate, NEGATIVE (B-IGNORE-A) • A device with two inputs typically labeled A and B, and one output that produces a logical 1 (true) when there is a logical 0 (false) at the B input, and a logical 0 (false) when there is a logical 1 (true) at the B input. The A input plays no role in determining output. If input A were always functional this would operate as a NOR gate, but in this case the A input can be disabled by a (temporary) signal. Upon receipt of this signal, input A is disabled, and this gate functions as a NEGATIVE (B-IGNORE-A) gate. When input A is disabled this gate functions like a NEGATION gate.

logic gate, NOR • The negation of an OR gate, (spoken as "NOR,") meaning "not or." A device with two inputs and one output that produces a logical 0 (false) condition when there is a logical 1 (true) present at either input.

logic gate, OR • A device with two inputs and one output that produces a logical 1 (true) when there is a logical 1 (true) condition present at either input or both inputs. OR is not an acronym, it literally means "or."

logic gate, XNOR • An exclusive NOR, or device with two inputs and one output that produces a logical 1 (true) condition when there is a logical 1 (true) condition present at both inputs, or when there is a logical 0 (false) at both inputs. Other input combinations cause an output of a logical 0 (false). XNOR is an acronym, as X stands for "exclusive," and NOR for "not or."

logic gate, XOR • An exclusive OR, or device with two inputs and one output that produces a logical 1 (true) when there is a logical 1 (true) at either, but not both inputs. The XOR produces a logical 0 (false) condition when there is a logical 1 (true) at both inputs, or a logical 0 (false) at both inputs. XOR is not a standard acronym, as X stands for "exclusive," but OR literally means "or." See acronym.

longitudinal • Situated or proceeding in a likewise or similar direction, as in a typically mounted automobile engine (in the same axis as the wheels of the vehicle). • In the physics of wave propagation (travel), a displacement of energy in the same axis (0 or 180 degrees), i.e. parallel to the direction at which a wave is propagating (traveling). Sound waves are longitudinal waves, and can be conceived as pulse(s) on a "slinky" (plastic or metal helical coiled spring). Cf. transverse.

lookup table • Digitized file in computer memory used in wave table-lookup synthesis, typically to produce a sound. The values of the waveform levels in a lookup table are scanned, or read at a selected standard playback rate to produce a specific frequency.

loudness • Subjective sensation of human hearing due primarily, but not exclusively to the intensity (power) of a sound heard. Many other factors influence human perception of loudness, particularly frequency, and to a lesser extent timbre and/or duration. The frequency of a partial (sine wave) obviously has a bearing on human perception of its loudness, because our hearing has frequency limits. Sounds outside of nominal human hearing limits (20 Hz – 20 kHz) are not heard at all, so a partial (sine waveform) near either frequency limit can't sound very loud, even when it is quite powerful (intense). This illustrates that signal amplitude is not necessarily equivalent to loudness, even in the case of an audio signal. The human ear is relatively insensitive to frequencies below 100 Hz, even those that remain within the audible window (20 Hz – 20 kHz), particularly at low intensity levels (small amplitude signals). Hearing is most sensitive near 3.5 kHz at any intensity level. See loudness curves, equal. Cf. volume.

loudness curves, equal • A set of shallow U-shaped "nested" curves that do not intersect or cross each other. Equal loudness curves relate various sound intensity levels (SIL) in decibels (dB) to various loudness levels (LL) in Phons (pronounced like "phones"). This Phon scale is a relative measure of perceived loudness versus signal intensity throughout the audible window (20Hz–20kHz). Curves for loudness levels (LL) spaced in 10 Phon increments, from 10-120 Phons are typically illustrated. By definition, decibel (dB) and Phon (LL) levels are equal at the reference frequency of 1000 Hz for all of the equal loudness curves. Equal loudness curves reveal that human hearing: (1) exhibits a frequency response that is more nearly flat at higher intensity levels (Phon curves at greater loudness levels are "flatter" than those at lower loudness levels); (2) is most sensitive at about 3.5 kHz at any intensity (dB) level; (3) is least sensitive at the extreme frequency limits of hearing, particularly at low frequencies produced at low intensity (dB) levels. These curves, or contours of equal loudness are aka *Fletcher-Munson or Robinson-Dadson curves after pairs of researchers who either *pioneered or replicated research that provides these data that illustrate relationships among frequency, intensity, and loudness based on human perception.

machine language • The lowest level, or most elemental form of commands used to operate a computer, consisting of signals that represent information symbolically as strings of zeroes (0) and ones (1). There are actually no zeroes or ones in a computer, only electronic signals or magnetic states that might be interpreted as such. Machine language is unlike high level languages (e.g. FORTRAN, LISP, BASIC, C++ etc.) that have alphanumeric commands (op codes) that execute many lines of machine language.

masking •

(math) multiplication • etc. (math) multiplication • etc.

matrix • A two dimensional rectangular array, or arrangement of data, items, quantities, or system elements in vertical columns and horizontal rows that optimizes access to and potential manipulation of such elements. A matrix in mathematics is usually treated as a single entity, and manipulated according to a particular set of rules. • The recording studio patch bay exemplifies a physical matrix, with inputs and outputs of remote or rack-mounted studio equipment routed, using cables, to the columns and rows on a panel of jacks centrally located within the studio. The patch bay locates inputs and output of remote, or "outrigger" equipment and recording console inputs and outputs conveniently in one area. See patch bay.

(measurement) accuracy • Quantitative measure of the extent to which a system's conditions conform to a known standard. A high degree of accuracy corresponds to a small amount of error. • For instance, a top octave synthesizer (TOS) is a chip, or electronic device that oscillates in the megahertz range, with divider circuitry that approximates, e.g. frequencies of the equitempered musical scale in the audible frequency range. The extent to which a particular TOS conforms to a particular specification of the equitempered scale, where musical half steps have a factor of the twelfth root of 2, i.e. ($2^{1/12}$) ≈ 1.05946:1 represents TOS accuracy. In context of this definition, it should be noted that the value for the twelfth root of 2 provided above would be more accurate if it were expressed using more digits, e.g. 1.059463094359295264561825294963 to the right of the decimal point. This "more accurate" value also provides greater precision (increased number of digits) and concomitant improved resolution (fineness of measurement due to reduction of step size). See (measurement) precision. See (measurement) resolution.

(measurement) precision • Quantitative measure that allows distinctions to be made between nearly equal values. An irrational number such as pi (π) expressed by six digits has better precision than one expressed by only four digits. However, an improperly computed six digit representation of pi may be less accurate that a properly computed four digit representation. That is, it is possible to express greater precision (e.g. six digits rather than four), but this does not necessarily ensure greater accuracy. The more precise representation may simply be in error with respect to some known standard of accuracy. Cf. (measurement) accuracy.

(measurement) range • Quantitative measure of the lowest and highest numbers, electrical values, or total bandwidth that a system can generate, modulate, process, or represent. • In mathematics, the set of values a variable or a function can express. • In music, the highest and lowest notes a voice or instrument can produce or the register of a passage of music from lowest to highest notes. • In music or sound synthesis, the extent to which a signal or signal usage, e.g. depth of modulation, represents a portion or percentage of the total possible range of possible values. • In general, system range and resolution have a relationship that inherently puts them at odds with each other. That is, when range is increased, given a fixed resolution, precision deteriorates, and accuracy is also likely to suffer. See (measurement) accuracy. The increment of the smallest possible step for numerical representations will grow larger as range is increased, given a fixed resolution. Resolution could be improved by restricting system range, a dubious approach (that has been done) to conceal low resolution in a system. Cf. (measurement) resolution.

(measurement) resolution • The extent to which a signal's accuracy and precision can be measured, determined, or resolved, expressed as a distance, percentage of full scale (FS), number of bits, or tolerance of error. For instance, the frequency resolution of a digital music synthesis system determines the accuracy with which one may generate notes in, e.g. an equitempered musical scale. See (measurement) accuracy. • Resolution also dictates the fineness of expressing a particular signal amplitude, level (size) or value, e.g. when attenuating a signal. • In digital systems, the number of bits (binary digits) used to represent signals constitutes that system's resolution. Resolution determines smallest available step size. Sixteen bit (16) sampling provides better resolution of sampled signal levels than does eight (8) bit sampling, as the ear attests. When resolution is low (too few bits), then both precision and accuracy likely will suffer when representing signals. • In general, system range and resolution have a relationship that inherently puts them at odds with each other. For example, when resolution is decreased, given a fixed range, precision deteriorates, and accuracy is likely to suffer as a result. The increment of the smallest possible step for numerical representations will grow larger as resolution is decreased, given a fixed range. Resolution could be increased by maintaining system resolution while decreasing the range, a dubious way (that has been done) to conceal low resolution in a system. Cf. (measurement) range.

(measurement) tessitura • The average pitch range, or most likely region of pitches performed by an instrument or voice in a particular musical passage, or more globally, in a complete composition.

(measurement unit) decibel • Logarithmic unit of measurement comprising a ratio of two signal powers, i.e. intensities. Cf. Bel. The decibel (dB) (as per ANSI standards) involves only comparisons of powers. A decibel is one tenth (1/10) of a Bel and is abbreviated dB (with no period). Because there are 10 dB per Bel, the formula for dB in sound intensity level (SIL) is: $SIL_{dB} = 10 \log (W_1 / W_0)$ with "W" in watts, a direct representation of power. See watt. Cf. Bel. Other letters such as I_1 / I_0 (intensities), or P_1 / P_0 (powers) are also used in the decibel (dB) and Bel formulas. The subscripts ($_1$) and ($_0$) in any of these cases are nominal, used as names that differentiate, such as "Sr." or "Jr." following a family surname. The decibel (dB) is known as a "dimensionless" measurement, i.e. one that does not necessarily require specification of a zero (0 dB) reference level. Two powers may be compared directly without specifying a zero (0) dB reference level. • When a particular 0 dB reference is specified, it is placed in the ($_0$) denominator of the dB formula, and the relative power of interest is then placed in the ($_1$

) numerator. See dB, zero. • Useful decibel facts: $10^{.301} \approx 1.9995$, which is approximately two (2), where ten (10) is our familiar number base. The superscript $^{.301}$ is the logarithm or log that represents .301 Bels, which is ≈ 3 dB (3.01 dB more exactly); and 1.9995 is the antilog, which is approximately equal to (≈) the power ratio of 2:1. Also: $10^{1.000} = 10$, where the first 10 is the base; the superscript $^{1.000}$ is a logarithm (1 Bel, or 10 dB); and the final 10 is the antilog, which represents a 10:1 power ratio exactly. From this we derive the memory aid, or mnemonic: "a doubling of power is 3 dB; ten times the power is 10 dB." The reader is exhorted to pound a heavy wooden table on the floor in common (4/4) time while shouting this mnemonic. See mnemonic. The initial downbeat is on "doubling," i.e. the word "a" is a pickup note. The word "ten" falls on the downbeat of the second measure. (Repeat ad nauseam.) By all means, do this with musical "style" (baggy pants with underwear generously revealed, baseball cap turned backwards, portable mic pointed skyward, etc.), and with rhythmic vigor (as your level of "musical talent" permits). We fully expect these musical desiderata to evolve over the years. • The decibel is named after Alexander Graham Bell (1847–1922), the Scottish scientist whose credits include invention of the telephone. Hence the upper case "B," in decibel (deciBel) and dB in honor of Bell, although this capitalization is not observed universally in either full length or contracted form. Cf. Bel. See dB, zero. • There is some evidence that Alexander Graham Bell's father, Alexander Melville "Biggie" Bell invented an elaborate notation system for rap music in 1867 (reference elsewhere: visible speech, a physiological language and phonetic notation system.)

(measurement unit) hertz • Frequency is expressed using the unit of measurement hertz (abbreviated Hz with no period), named after Heinrich Hertz, an early physicist and radio experimenter. The older unit of measurement cycles per second (cps) may be more descriptive, but it has been superseded by the Hz unit. • Frequency (f) is the inverse of period, the time (T) required for one cycle of a periodic waveform to complete, that is: ($f = 1 / T$). And, as principles of math dictate, period (T) is necessarily the inverse of frequency (f), that is: ($T = 1 / f$). See relationship, inverse. See period.

memory • The subsection in a digital computer capable of storing data or commands, typically represented by combinations of bits. See bits. See (memory) etc.

memory, cache • High speed computer memory designed to temporarily hold the most recent data accessed from some main storage device, e.g. a disk drive, in order to accelerate memory access time. • Pronounced like "cash."

(memory) EPROM • Acronym for electrically programmable read only memory (EPROM), a read only memory (ROM) that can be "burned" or programmed only once, after which its contents cannot be altered. See (memory) ROM. • Pronounced "E´PROM."

memory, flash • Form of random access memory that does not require a power source to retain memory contents. See (memory) RAM.

(memory) RAM • Acronym for random access memory (RAM), a type of computer memory that allows storing and accessing data in memory without recourse to sequential or serial movement through numbered memory addresses. That is, any available RAM address can be accessed directly, without considering its numerical position in an array or matrix of memory locations. Magnetic tape used as a data storage medium for early digital computers exemplifies serial rather than random access to memory addresses. • RAM is considered to be in contradistinction to read only memory (ROM), which does not allow data storage, or "write" operations to memory addresses by the user. That is, data are permanently "burned" or fixed into ROM by the designer. • The designation random access memory (RAM) misdirects attention from the fact that both RAM and ROM provide random access to any available memory address. Volatility, i.e. whether memory loses its contents when power is turned off, is actually the chief distinction between RAM and ROM, not the manner of reading memory addresses. RAM is volatile, meaning it loses its memory contents when not supported by electrical power. ROM is non-volatile (retains memory contents), and therefore needs no power source to "hold up," or retain data written to its addresses. • RAM might more aptly be called "volatile read-write memory," but "VRWM" isn't a neat acronym that is easily pronounced! Cf. (memory) ROM. • The appearance of solid state non-volatile memory technology bids to make prior distinctions about memory volatility essentially of historical interest. See memory, flash.

(memory) ROM • Acronym for read only memory (ROM), a type of computer memory that allows only "read" operations from its memory, but no "write," or storing of user-generated data to such memory. Both ROM and so-called "random access" memory (RAM) provide random access to memory locations. That is, neither type of memory requires sequential, or serial movement through numbered memory addresses to locate a specific address. Any memory address may be accessed directly by numerical location. • ROM is named appropriately, but its name does ignore the concept of volatility. ROM is non-volatile (retains memory contents) and therefore needs no power source. ROM might be named non-volatile read only memory to better describe a major distinction from RAM. • Cf. (memory) RAM.

meter • Device or instrument that senses, measures, quantifies, and displays some specific characteristic of a signal using analog or digital means. • Pattern of successive musical pulses or loudness emphases that represents the progression and subdivision of (relative) musical time, e.g. 3/4 or 4/4 time. These fractional representations are aka time signatures in music. The time signature ¾ indicates that 3 quarter notes will constitute a "measure." • An internationally accepted (SI, or Système International d'Unités) standard for length, equal to approximately 39.37 inches in the English system of measurement. See metric system.

meter, peak • A peak meter displays instantaneous signal level that exceeds a given threshold, in contradistinction to an averaging meter such as the VU meter. Typically, a peak meter enables (turns on) a single light emitting diode (LED) or other visual indicator when a monitored signal exceeds the predetermined peak (highest) level of the system, indicating onset of unacceptable distortion. A peak meter is a threshold detector. See threshold detector. Cf. meter, VU.

meter, VU • A volume unit (VU) meter shows the average signal level over a fixed time constant, in contradistinction to a peak meter, which responds more quickly to indicate a signal that exceeds some predetermined full scale (FS) value. Cf. meter, peak. The physical mechanisms of a hardware VU meter must overcome inertia, giving rise to so-called meter "ballistics" that smooth, or average (integrate) more-rapid signal fluctuations. Take note that a VU meter depicts, when displaying a 440 Hz sine wave, not the actual rapid alternations of that waveform's many positive (+) and negative (–) values, i.e. the actual "micro" changes this selected waveform exhibits. The VU meter depicts, rather the "macro," or "steady" average amplitude of this (and other) waveform(s). Software-based VU meters use mathematics to mimic, i.e. physically model this physical inertia, or directly compute its transfer function, in order to provide signal averaging that mimics the action of a hardware VU meter.

metric system • System of weights and measures based on the decimal (base ten) system, using the meter (length), the kilogram (weight), and the second (time), rather than arbitrary or historical standards, such as the length of some randy English King's foot. The metric system is predicated on powers of the decimal number system with its base, or radix of ten (10). Rational prefixes such as deka (10^1), hecto (10^2), kilo (10^3), mega (10^6), giga (10^9), tera (10^{12}), peta (10^{15}), exa (10^{18}); and deci (10^{-1}), centi (10^{-2}), milli (10^{-3}), micro (10^{-6}), nano (10^{-9}), pico (10^{-12}), femto (10^{-15}), and atto (10^{-18}) are used to indicate (positive or negative) powers of the base ten (10) respectively. The modern version of the metric system embraces the SI (Système International d'Unités) system of units, which enjoys widespread international agreement. • Metric system prefixes based on powers of 10 are rational, but people are not necessarily so. Consequently, metric system prefixes are sometimes used irregularly. For example, a "kilobyte" might reasonably be expected to be 1000 bytes, as kilo is a specific (10^3) power of 10 that represents one thousand (1000) of something. But a kilobyte is actually 1024 bytes, a power of two (2^{10}), due to the binary (rather than decimal) basis of modern digital computer technology. Similarly, 64 "k" (kilo, implying 10^3), e.g. the number of different levels available in 16 bit PCM sampling, refers not to 64 000 (64×10^3), but to 65 536, which is (2^{16}). The intended number in context of digital computer technology is actually a power of two (2), rather than ten (10), even though nominally metric prefixes are used. In his definition for "byte," White (2005) wryly notes: "Learning such irregularities is one of the keys to becoming proficient in the domain of digital systems and computers." Budding music technologists should be aware that there are so many "irregularities" in the field of music technology, that one might lose sight of the original mission: to make art.

metronome • Electronic or mechanical device that produces a periodic audible and/or visible signal that represents a particular subdivision of absolute time, e.g. a musical tempo in beats per minute. Older mechanical devices have a mechanical clockwork system comprising an inverted pendulum upon which is mounted a movable weight, all of which is driven by energy stored by a tightened coiled spring. Modern metronomes embody electronic timing circuitry. Most metronomes offer a selection among metronomic markings (MM) calibrated in beats per minute. ”MM = 88” means that 88 beats, or pulses will occur per minute. Sophisticated electronic metronomes may provide subdivisions of beats using different sounds. See tempo.

microphone • Transducer that features a mechanical element, or diaphragm that vibrates due to impinging kinetic energy such as sound waves and changes these vibrations into potential energy (electrical signals, e.g. voltage) at its output. Sometimes shortened to “mic” (rhymes with “like.”) “Microphone” literally means “small voice.” See transducer.

microtonal • Musical scale with frequency subdivisions smaller (after “micro,” meaning “very small”) than the traditional dodecaphonic (12 tones per octave) scale. “Microtonal” is deemed by some to be restricted to a musical scale whose smallest interval (e.g. one-quarter tone) has a consistent single frequency ratio, producing a scale with intervals of equal size, in contradistinction to various temperaments, in which nominally similar intervals (e.g. half steps) may actually vary slightly in size, i.e. have slightly different frequency ratios). That is, any nominal interval, e.g. half steps in a temperament system may not be precisely equal in size, whereas in equitempered scale(s) or temperaments all intervals of a given nominal size (half steps in this example) have precisely the same frequency ratio (intervallic size). See interval, musical.

MIDI • Musical Instrument Digital Interface (MIDI), a serial communications protocol that debuted with a bit rate of 31 250. (The MIDI Manufacturers Association (MMA) recognized the need to improve and extend this protocol in 2006–2007 but results remain to be seen). The first version (1.0) of MIDI is a “gesture-based” protocol, having no particularly compelling or even necessary relationship with audio or music per se, but MIDI can communicate the actions of a musical performer, particularly actions performed using a standard keyboard. See baud.

mix • To combine and process multiple tracks of audio signals in order to create an audio playback experience or performance using a fixed number of playback channels, e.g. mono, stereo, 5.1 surround, etc. • Signals other than audio can be summed, terminology the authors prefer to “mixed,” a term generally understood to involve audio signals. See summing node. Cf. mixer. • The outcome of actions that create a mix of audio signals, one of many possible mixes that might be made–a finalized product.

mixer • Hardware or DSP capability that facilitates summing, scaling, distributing, and processing signals. The term is commonly associated with means designed to process broadband audio signals, although a mixer can process signals other than audio if its design permits. Cf. summing node. • Typical mixer features are replicated on each channel strip of an audio engineering mixing console. See mixing console. • The most elemental part of a mixer comprises circuitry or DSP capability that constitutes a summing node or summer. A summing node provides no scaling, or level control, only signal summation. A summing node is found within each signal controlled module, e.g. SCO (signal controlled oscillator), SCF (signal controlled filter), SCA (signal controlled amplifier), etc. It algebraically adds (±) signals connected to internal and external control inputs, to facilitate simultaneous control of the selected module parameter by more than one signal. See algebraic addition. For example, SCO frequency might be controlled simultaneously by coarse and fine tuning constants, keyboard note number, and LFO (low frequency oscillator) signals. In this case, the level of each applied control signal is scaled (attenuated) individually using an associated attenuator that is not part of the summing node per se. See summing node.

mixing console • Broadband hardware or virtual means designed to process audio signals, providing signal summing and other signal processing capabilities: (1) gain control, e.g. fader, pad; (2) spatial placement of channel information, e.g. stereo or multidimensional panning; (3) equalization, i.e. various filtering; and (4) signal routing, e.g. various signal busses such as send, receive, monitor, etc. A mixing console or mixer typically has more inputs than outputs, expressed as, e.g. "8 into 2," and is used to combine many tracks of audio signals in preparation for recording, and/or presentation in a smaller number of playback channels found in a standard recording/playback format such as stereo, 5.1 surround, 7.1 surround, etc. See fader. See panning. See EQ. See gain. See mix. See mixer. Cf. summing node. • A mixing console is also known in the recording studio as a "desk," or "board."

mnemonic • A memory aid (pronounced nuh mon´ik), often expressed as an acronym, catchy phrase, saying, rap, or rhyme intended to be memorable. For instance, "every good boy does fine" is a mnemonic for the notes E-G-B-D-F that appear successively from lowest to highest "lines" of G-clef, aka treble clef on the Grand Staff (ten horizontal lines displayed in two groups of five each) used as the standard for traditional music notation. In this context, the word "face" spells out the notes F-A-C-E in the "spaces" between the lines of treble clef (G-clef). Cf. acronym.

modem •

modulation • Any of a number of techniques that use a carrier (C) signal, one of whose signal characteristics is changed, or modulated by an associated modulation (M) signal. See (modulation) etc. • More generally, a change, particularly a repeating (typically periodic) change of a parameter of a signal in a sound synthesis system that may result in, e.g. a low frequency modulation such as vibrato (FM) or tremolo (AM). See tremolo. See vibrato. See (modulation) AM. See (modulation) FM. • In music, a change from one key to another, particularly in cases where the keys are related, e.g. Major to relative minor (e.g. G Major to e minor).

(modulation) AM • Amplitude modulation (AM) alters the amplitude of a carrier (C) signal, based on the waveform characteristics of a modulation (M) signal. In sound synthesis, AM involves listening to the carrier, unlike AM radio, where the modulation signal embodies the information of interest (music, speech). Each channel of AM radio requires a carrier frequency far higher than the upper limit of human hearing, whose amplitude is modulated by the modulation signal (music, speech). This carrier signal is "demodulated" in order for us to actually hear the music or speech that the modulation signal embodies. • Classic, or "unbalanced" AM may be effected using a module called a two quadrant multiplier, known in early analog systems as a "voltage controlled amplifier" (VCA, now SCA "signal controlled amplifier.") See multiplier, two quadrant. See VCA. Cf. multiplier, four quadrant for "balanced" AM. See (multiplier) BAM. • In sound synthesis systems, sonic outcomes of AM depend largely on the signal characteristic(s) of the modulation signal (M), given a carrier (C) signal that has a frequency in the audible window (20 Hz – 20 kHz). AM technique can produce: enveloping (infrasonic, aperiodic M signal, e.g. from an envelope generator); tremolo (2–7 Hz periodic M signal, e.g. from an oscillator); and production of sideband frequencies (high frequency periodic M signal, e.g. from an oscillator). • Classic rapid modulation AM sideband production uses a two quadrant multiplier (VCA/SCA) to multiply the carrier (C) and modulator (M) signals, both of which are typically audio frequency periodic waveforms. Given audible frequency sine waves at inputs (C) and (M), with a unipolar positive M signal, AM can output the original carrier (C) signal, as well as two other partials, a "sideband pair." See sideband. This AM sideband pair comprises partials at C ± M (C plus M, and C minus M) frequencies for each partial in the carrier signal. The two partials in a sideband pair have the same level, which does not exceed one half (½) the level of the carrier (C), in classic AM. The modulator (M) input signal is suppressed in AM, i.e. it does not appear at the multiplier's signal output, and is therefore not heard. • AM is a nonlinear process, i.e. one that can produce (potentially many) partial(s) whose frequencies were not originally present in either of the (C or M) input signals. Therefore, AM is a valuable technique capable of producing the non-harmonics that appear in "klang" (bell-like) sounds. See multiplier. Cf. (modulation) BAM.

(modulation) BAM • Balanced amplitude modulation (BAM), aka "ring" modulation is a variant of amplitude modulation (AM). See (modulation) AM. BAM uses a four quadrant multiplier to suppress both carrier (C) and modulator (M) input signals. Only a sideband pair at C ± M (C plus M, and C minus M) input frequencies results from BAM when there is a sine waveform at both C and M inputs. Both inputs in a four quadrant multiplier accept bipolar (±) signals; these inputs are sometimes designated simply as "A" and "B" rather than "carrier" and "modulator," as choice of input for signals is moot, unlike the situation with a two quadrant multiplier. See multiplier, four quadrant. Cf. multiplier, two quadrant. • BAM is a nonlinear process, i.e. one that can produce partial(s) whose frequencies are not present at the processor's input(s). See multiplier, four quadrant. Cf. (modulation) AM.

modulation, bi-phase

(modulation) DM

(modulation) FM

modulation, half wave

(modulation) PAM

(modulation) PCM

(modulation) PDM (PWM)

modulation, phase

(modulation) PPM

(modulation) PWM

modulation, ring • See (modulation) BAM. In early analog circuit designs that provide balanced amplitude modulation (BAM), a circle, or "ring" of connected diodes is featured prominently as part of the circuitry. See diode.

modulation, sigma-delta

modulation index • See FM modulation index.

modulation sidebands • Partials, or sine waves that appear in pairs at integer, or positive (+) and negative (–) whole number multiples of each individual partial's frequency in the modulation signal, due to various modulation or sampling operations. That is, e.g. in the case of classic linear FM, where both carrier and modulator signals are sine waves, sidebands occur at whole number multiples of the modulator frequency above and below the carrier frequency. Music synthesis techniques such as amplitude modulation (AM), balanced amplitude modulation (BAM), and linear frequency modulation (FM) can produce one or more sideband pairs of sine waves per each sine wave (partial) in a complex wave carrier (C), given a sine wave modulator. See FM. See (modulation) FM. See (modulation) AM. See (modulation) AM. See BAM.

module • Hardware unit or virtual equivalent in a sound synthesis system in which individual units can be selected, removed, rearranged spatially, and connected using patch cords or other means, to effect interactions among modules that create different audio, control, and/or timing structures or subsystems. Each module is usually self-contained, and performs a specific role within the system, e.g. signal generation, signal processing, timing, terminal operations such as input/output, etc. A given module is categorized as a generator or a processor. See (module) generator. See (module) processor. See ("module") terminal. • Modules are often represented graphically using simple line drawings called icons.

(module) generator • One of two classes (generator or processor) of modules in a modular sound synthesis system, a generator produces a signal. A generator has no signal input, it has a signal output. A generator may or may not have timing (T), and/or control (C) input(s). Cf. (module) processor. Cf. (module) terminal.

(module) processor • One of two classes (generator or processor) of modules in a sound synthesis system, a processor alters (processes) a signal. A processor has a signal input, as well as a signal output, and signals can pass through it. A processor may or may not have timing (T), and/or control (C) input(s). Cf. (module) generator. Cf. (module) terminal.

("module") terminal   • In the authors' schema, a terminal is a device or capability that provides a means for routing signal(s) into or out of a virtual (software-based) system or subsystem. In the most general sense, a terminal is a port, or set of ports, that stand on the "frontier" of a system. A terminal involves input or output from or to a virtual system or subsystem. An external signal routed into a virtual system may ultimately function as an ACT (audio, control, timing) signal, but the (input) terminal itself is not an ACT module–it is only a conduit (pipe, tube, wire, or other device that conveys information, a signal, or something else conceived to flow). That is, the function of a terminal is neutral, and remains undefined until connected to the outside world. Within the virtual system any module is either a generator or a processor, and has no function prior to its connection to an ACT input in the internal (virtual) world. See ACT. A signal routed out of the virtual system might ultimately function in those (ACT) ways, but the terminal through which it is routed is strictly a conduit, not a functional ACT (audio, control, timing) module. Even an output nominally labeled "audio" does not necessarily have to be connected to an audio system. Although it may be represented by an icon that looks like a module, a terminal is neither a generator nor a processor, so does not fall within our binary classification of module types. Some terminals include processing capabilities, e.g. gain control. In such a case, it is the gain control that is defined as the processor, not the terminal per se. See ACT.

monitor   • Hardware used to hear or view mediated electronic or digital information, e.g. loudspeakers for playing audio signals, and video display terminals (VDT) for displaying images produced by a computer, DVD player, television transmission, etc.

monophonic   • Sound system in which any signal played will provide the same information to all loudspeakers used. That is, a monophonic audio system has a single channel to communicate from signal source (computer hard drive, playback mechanisms, CD player) to the loudspeaker(s). Cf. stereophonic. • Musical instrument designed to sound one tone or play one key (keyboard) at a time. Cf. polyphonic.

multiple   • Passive hardware configuration or freestanding module with several jacks, any one of which can act as an input, thereby making several copies of that input signal available at all the other jacks in that module, which subsequently act essentially as signal outputs. A multiple, abbreviated "mult" (with no period), provides a way of creating clones, or exact replicas of a signal. When more than one signal is routed into a multiple (not its "intended," or conventional use), summation (mixing) of these signals is nonlinear, with results that are not easily predictable. (A multiple is not designed to be a summing node or a mixer, either of which provides linear, algebraic (±) summation of signals.) See algebraic addition. See device, passive. Cf. mixer. Cf. summing node. • In most virtual modular sound synthesis systems, the need for a multiple module is vestigial (no longer necessary, required, or useful), as such software systems typically allow simultaneous connection of any signal output to any number of inputs in the system. (A virtual system can create as many "clones" of an output signal as warranted, without recourse to a dedicated multiple module. That is, a given output can typically be connected to any number of inputs as needed.)

multiplexing • Any of a variety of techniques that facilitate the capacity of a hardware or software system to simultaneously transmit and/or receive an increased number of individual, separate messages on a single virtual channel, radio or microwave transmission frequency, or hardware cable or wire. Multiplexing provides means of carrying many, e.g. telephone conversations on a single wire, or on a single microwave carrier frequency. Among various means of multiplexing are frequency division and time division techniques. See (modulation) etc.

multiplier • Processor that multiplies the instantaneous signal level at one input times the instantaneous signal level at its other input and provides the mathematical product of these two input levels at the signal output of the module. In DSP terms an amplifier is a multiplier. See amplifier.

multiplier, four quadrant • Processor that multiplies the instantaneous levels, (numbers in DSP terms) at its two inputs. These inputs do not necessarily make distinctions between "carrier," and "modulator," in which case the inputs may be simply labeled "A" and "B." Cf. multiplier, two quadrant. A four quadrant multiplier accepts a bipolar (±) signal at both "A" and "B" inputs. That is, a four quadrant multiplier multiplies either negative or positive "A" values by either negative or positive "B" values, and outputs a value of zero (0) only when either "A" or "B" input signal has a value of zero (0), as per principles of multiplication. See (modulation) BAM. • The "four quadrant" designation (I, II, III, IV) refers to a two dimensional Cartesian plane or grid that accommodates (x,y) coordinate pairs, i.e. points that can be used to depict multiplier "A" and "B" numbers (or signals) on the same graph. (In the case of audio frequency amplitude modulation (AM), neither "A" nor "B" input signals appear at the signal output of a four quadrant, aka a "balanced" multiplier. That is, both "carrier" and "modulator" are suppressed by a four quadrant (balanced) multiplier. Only the C+M and C–M sideband pair produced by this process appear at the multiplier signal output. See (modulation) AM.) Because both carrier and modulator signals accept bipolar (±) values, all four quadrants in the Cartesian plane are potentially involved, depending on the polariti(es) of the input signals. Cf. multiplier, two quadrant. See (x,y) values.

multiplier, two quadrant • Processor that multiplies the instantaneous levels, or in DSP terms, the numbers at its carrier (C) and modulator (M) inputs. See multiplier. A two quadrant multiplier has a carrier input that accepts bipolar (±) signals, and a modulator input that accepts only unipolar positive (+) signals. That is, a two quadrant multiplier cannot multiply the carrier signal times negative modulator values, and subsequently outputs zero (0) when the modulator is zero (0) or negative (–). Of course, if either carrier or modulator value is zero (0), output will be zero (0), as per rules of multiplication. • The "two quadrants" occupy half of a two dimensional Cartesian plane or grid that displays (x,y) coordinate pairs used to describe both carrier input and carrier output numbers or signals on the same graph. (Modulator signals (M) do not appear at the signal output of a two quadrant multiplier. Only carrier (C) signals can appear at the signal output of a true two quadrant multiplier.) Because the modulator input is constrained to accept only positive (+) values in a two quadrant multiplier, only two of the four quadrants in the Cartesian plane are used. The four quadrants (I, II, III, IV) refer to a two dimensional Cartesian plane or grid that accommodates (x,y) coordinate pairs to map two variables or values. See (x,y) values. • The classic Model 902 voltage controlled amplifier (VCA) in Robert A. Moog's 900 Series modular, voltage controlled synthesizer (1964) is a two quadrant multiplier. In virtual designs, the designation signal controlled amplifier (SCA) is probably an appropriate new convention, and appropriate for a two quadrant multiplier. Cf. multiplier, four quadrant. It should be noted that many contemporary designs for virtual modular synthesis systems make no distinction between two and four quadrant multiplier modules, in some cases providing only the latter.

(music) heterophony or heterophonic • Music exemplified by playing and/or singing essentially the same line, or several improvised or ornamented versions based on the same melody, in several voices in parallel (simultaneously). Heterophony may occur naturally when men, women, and children sing essentially the same melody at intervals of the octave and (sometimes) perfect fifth (P5) due to natural differences in the average pitch registers of the performers. • Performance in which a melody is sung, accompanied by an instrumental performance of an ornamented version of the melody. • This means of making music is indigenous to parts of Asia, West Africa, and the Near and Middle East. Cf. (music) polyphony.

(music) homophony or homophonic • Music that has a melody centered in one voice with a simple accompaniment that plays a subordinate role, as in modern Protestant hymns. When all parts move with the same or similar rhythms, the texture is called homorhythmic. Cf. (music) polyphony.

(music) monophony or monophonic • Music that consists of an unsupported melody, or single "voice." Examples in Western music start with plainsong, and include music of the troubadours, minnesingers, meistersingers, etc. The folk song, while monophonic in principle, may often be sung with an improvised accompaniment. See monophonic. Cf. (music) polyphony.

(music) polyphony or polyphonic • Music that features several lines, or "voices," each of which maintains its musical identity. Polyphonic lines are often referred to as constituting counterpoint, and their use, particularly in structural forms such as the fugue, is deemed contrapuntal music. See polyphonic. Cf. (music) monophonic.

musique concrète • (mu zeek´ kon kreht´) Literally "concrete music," in contradistinction to what might be called "abstract music," i.e. traditional tonal or atonal music played by performers using musical instruments in order to realize compositions based on music notation. The term musique concrète was coined by Pierre Schaeffer to describe his celebrated "concert of noises" that appeared on ORTF (RTF) radio in Paris, France in 1948. Musique concrète uses recordings of real world sound(s), historically using disks initially, evolving to analog magnetic audio tape, and now using digital audio recordings. Recordings using any such media are transformed or manipulated using so-called "tape" techniques (a vestigial, but descriptive term) that have largely been superseded by homologous (functionally similar) digital signal processing (DSP) operations. These techniques include audio reversal, alterations of recording vs. playback speeds (pitch shifting), tape splicing (playlisting or editing), montage (overdubbing), looping, splicing, etc. Electronic processing using filters, ring modulators, amplifiers, attenuators, frequency shifters, etc. may be involved as well. Cf. elektronische musik. • The original genre distinctions between musique concrète and elektronische musik, a product of the nominally "first" (ca. 1950) electronic music studio located in Köln (Cologne), Germany are essentially of only historical interest, as techniques derived from both modalities of composition and sound production remain in widespread use today.

mute • Button or switch on each channel strip of an audio recording console that toggles (alternates) between silencing or enabling that channel's contribution to the final output or intended sub-mix. Cf. solo. • Device for various orchestral instruments that causes a reduction of loudness, and a concomitant change of timbre.

muting • Digital error correction technique that sets the value of missing or uncorrected data in digital audio to zero (silence), thereby avoiding audible, unwanted sound artifacts, i.e. distortion. • Actions involving use of mute controls on a recording console. Muting a channel causes its sonic contribution to not be heard on the bus or present mix that is operational. See mute.

negative amplitude • Amplitude with a signal polarity (–) opposite (180°) that of an amplitude deemed or determined to be positive (+). See amplitude.

negative frequency • Frequency with a signal polarity (–) opposite (180°) that of a frequency deemed or determined to be positive (+). See frequency.

noise • In general terms, noise is any sound or signal that is unwanted, particularly unwanted signal(s) that intrude upon desired signals in a telecommunications or audio engineering system. In such a context, noise might also be defined subjectively–one person's noise is another person's music. • In objective, technical terms broadband noise is a random signal that has a particular signal amplitude vs. probability distribution (e.g. Gaussian), as well as a specific frequency vs. amplitude characteristic (e.g. white noise, pink noise (1/f noise), $1/f^2$ (Brownian) noise, etc.) Such signal(s) may be deemed desirable and are often used in music and sound synthesis for artistic purpose. • A random signal has a waveform (variations of level in time) that has no specific period, no repetitive pattern, and no predictable relationships among instantaneous appearances of a particular instantaneous waveform level over time. Amplitude excursions of white noise have a gaussian or "bell-shaped" probability, meaning that the most extreme excursions are less likely. • Broadband random noise, or Gaussian "hiss" has afflicted analog audio recording media of all kinds. Digital audio systems are essentially immune to such broadband noise, but can suffer from quantization noise when the number of bits used for quantizing (sampling) is insufficient. See noise, quantization. See noise, etc.

noise, $1/f^2$ • Inverse square frequency noise ($1/f^2$), spoken as "one over f square," is a random signal, called variously "Brownian," low frequency, or "red" noise. The frequency vs. amplitude distribution of $1/f^2$ noise exhibits inverse square levels with respect to frequency, i.e. the lower the frequency, the higher the power. A given frequency such as 100 Hz that is one half that of another frequency (200 Hz), has 1/4 (1/2 squared) the power of the higher frequency (a 6 dB difference). Inverse square frequency noise, or $1/f^2$ noise came to attention due to the burbles and bumps (noise) present in any signal involving semiconductor (electronic component) devices, caused by spontaneous generation and recombination of sporadic currents. The term "Brownian motion" came into common use due to studies of atomic physics that reveal a "random walk" of small particles.

noise, broadband • (Typically) refers to a random signal, e.g. pink or white noise, that covers the range of human hearing, from 20 Hz–20 kHz. See noise, pink. See noise, white.

noise, pink • A broadband random signal with partials at all audible frequencies, with random fluctuations of phase and amplitude per each partial, aka "1/f" (spoken "one over f" noise). See noise. When averaged over time, pink noise exhibits equal power per equal ratiometric frequency bandwidths, i.e. equal musical intervals. For example, the octave bands between 1 kHz–2 kHz and 10 kHz–20 kHz have the same power in pink noise. Cf. noise, white. • Pink noise is used for many tests in audio engineering. Unlike white noise, the sound of pink noise is not dominated by highs, but has a more apparently flat tonal balance, due to the near logarithmic response of the human ear. Each successive higher octave in pink noise has been equalized, conceptually (or literally) by passing white noise through a –3 dB/Octave low pass filter. Pink (1/f) noise has a "flat" frequency vs. power distribution per equal musical interval, e.g. octave, ½ octave, 1/3 octave, 1/6 octave bandwidths. Pink noise is not flat per equal linear frequency bands (in Hz), exhibiting a relative loss per progressively higher bands of equal frequency band size, e.g. 1–2 kHz, 2–3 kHz, 3–4 kHz . . . 19–20 kHz. A linear frequency display illustrates this as a frequency response curve with a slope exhibiting a three decibel loss per each successively higher octave (–3dB/Octave). The designation "1/f" for pink noise means that a given frequency that is half as high as another (100 Hz compared to 200 Hz) has half (1/2) the power (–3 dB) of the higher frequency. See filter slope. Cf. noise, white.

noise, quantization • The residual error component, or math term that remains when a properly sampled digitized signal is compared to the original analog signal from which it was created. Quantization noise, aka quantizing noise or quantizing error comprises the mathematical elements, and consequently the sampled signal artifacts that remain following subtraction of the digitized (sampled) signal from its original analog form. This quantizing noise is part of the sampler output. Because an analog signal has a theoretically infinite number of possible levels, and any digital representation, or sampled version of an analog signal necessarily has a finite number of possible levels, error (noise) due to the sampling process is inherent. In practical terms, a higher number of bits reduces quantization noise (the errors) to a negligible level. A general rule of thumb states that quantization noise, or quantizing error increases by approximately +6dB per each decrease by one (1) of the number of bits used in sampling. For example, 8 bit sampling is inherently 48 dB (6dB x 8 bits) “noisier” in terms of quantization error than is 16 bit sampling. • It is important to make distinctions, both intellectually and aurally, between quantization noise and broadband noise. Broadband noise (typically with Gaussian, or “bell shaped curve” probability distribution for amplitude excursions) includes all audible frequencies, as exemplified by the ever-present “hiss” of analog magnetic tape recordings. Tape hiss occurs essentially due to the residual random dispositions of polarities of a significant number of magnetic domains (comprising tiny metallic oxide fragments) on tape after recording. This explains why an analog magnetic tape that is fully “saturated” due to recording a program signal at a high level produces less (broadband) noise. Objectively, fewer magnetic domains on the tape exhibit random magnetic polarities following recording, physically reducing the amount of noise actually present on tape. Subjectively, the residual noise that remains on the tape is masked by the much larger program signal, making it difficult to perceive the noise that is actually present in the playback signal at some residual level at all times. Typically, when the program (recorded signal) falls to a low level, the resident “noise floor” is easily heard. See noise, white. Quantization noise, a digital artifact endemic to sampling, is a band-limited signal, unlike broadband analog tape noise, or “hiss.” Audible quantization noise has a different sound than broadband noise, one that becomes highly irritating to the nerves–creating a certain “grittiness” or “grunge” in the texture of the recording (technical terms, to be sure.)

noise, thermal • Noise due to a basic nature of physical atomic structure, particularly when matter is heated, thereby making atomic particles tend to undergo random movements that create electrical disturbances with equal energy per equal bandwidth of frequencies, e.g. 1 kHz–2 kHz and 10 kHz–11 kHz. Such an equal energy per bandwidth noise distribution is aka white noise by analogy to white light, which includes all frequencies of visible light. See noise, white.

noise, white • In audio terms, white noise is a broadband random signal with partials at all audible frequencies, with random fluctuations of phase and amplitude per each partial. “White” noise is so-named by analogy to white light, which encompasses all frequencies of visible light. When averaged over time, white noise has equal power per equal linear frequency band, i.e. per equal spans of frequencies in hertz (Hz). For example, frequency bands at 1 kHz–2 kHz and 10 kHz–11 kHz have the same power, as does any other inclusive band with a similar 1000 Hz bandwidth. • A white noise audio signal is dominated by “highs,” because successively higher octaves exhibit a progressive doubling (2:1 ratio) of power, yielding a +3dB increase per each successive higher octave. (A doubling or halving of power is a three decibel change). To illustrate this, let’s create a thought exercise: imagine that frequencies are restricted to whole numbers, and it then becomes apparent that the band of partials between 2 kHz–4 kHz has “twice the number” of partials than the band between 1 kHz–2 kHz, accounting for the doubling of power of the higher octave. Similarly, 4 kHz–8 kHz has “twice the number” of partials than 2 kHz–4 kHz, and so forth. Successive higher octaves are therefore +3dB (two times) more powerful per octave. (Frequency resolution of white noise is not restricted to whole numbers of course, but this thought exercise is revealing, and the associated reasoning is valid in principle.) See frequency response, flat. Cf. noise, pink.

noise floor • The level of noise continuously present in a circuit or device that contains electronic components or electromagnetic elements. This level, which is typically stated in decibels (dB), is present due to the noise inherent in all electronic devices. See noise, thermal. For instance, the S/N (signal to noise) ratio of a system is determined by the power ratio of the level of the largest allowable signal that produces "tolerable," meaning near-zero distortion, compared to the level of the resident noise floor signal measured when there is no input signal present. The noise floor is consequently reported as some negative dB (decibel) level "below" the largest allowable signal. See S/N ratio.

noise gate • Broadband processor that senses whether the instantaneous level of an input program audio signal falls above a programmable threshold level, or not. While above the threshold setting, this program signal (e.g. music, speech) passes directly through the noise gate (NG) to its signal output, essentially bypassing this processor. Any program level that does not exceed the threshold causes the noise gate to quickly "gate," i.e. attenuate the program signal, reducing its output level to zero. This completely silences the program signal so it is consequently not heard. Complete (near 100%) attenuation is not instantaneous, but occurs in the programmable time of a release (aka decay) envelope. When the instantaneous program level once again exceeds the threshold, the noise gate turns its output back on, in the programmed time of an attack envelope. • Why use a noise gate (NG)? Because signal to noise (S/N) ratio deteriorates, expressed as a smaller S/N ratio in decibels (dB), when the level of a program signal, i.e. what you want to hear, becomes smaller relative to: (1) the noise level continuously present in the system (noise floor); and/or (2) any background or residual noise in the recorded program signal itself. A program signal with an instantaneous level that falls on or below the programmed threshold is not passed by the noise gate at all, thereby silencing quieter (low level) passages of the program that otherwise would have a lower (worse) S/N ratio than louder passages. No signal is output from the noise gate at all when the input signal falls below the threshold level. A noise gate turns off, or silences both program and noise signal(s) when the program signal level falls on or below this typically adjustable threshold value. • An archetypal need for a noise gate exists when music or speech (program signal) is recorded in the presence of unavoidable ambient noise, e.g. motor vehicle traffic. This is a chronic audio problem encountered in electronic news gathering (ENG) for broadcast. A noise gate does not remove noise, neither internal system noise nor noise in the program signal, when the noise gate passes the program signal connected to the signal input of the NG. It passes the desired program signal as well as both kinds of noise, internal system noise as well as any ambient noise that is part of the (recorded) program itself. The utility of a noise gate is based on the phenomenon that larger program signals, e.g. the louder portions (e.g. speech sounds of the interviewee) that are passed, tend to mask (cover up) both kinds of noise that are actually still present in the passed signal. Then, when a recorded interviewee is not speaking, e.g. at the pause at the end of a sentence, there is then no masking program signal whatsoever, and the ever-present internal, and any recorded ambient noise signals (e.g. traffic noise) would become more apparent if the noise gate were not used. So, because all output is stopped by the noise gate during such intermittent low levels in the program signal (e.g. when an interviewee falls silent), our awareness of both kinds of noise is reduced. The audio engineer must judge where to set the threshold of a noise gate, and the respective times of attack and release envelopes, to optimize intermittent use of the noise gate, while minimizing degradation of the desired program content. If the input program threshold is set badly, and/or attack and release envelope times are aberrant, the "pumping" or "breathing" action of the noise gate itself will become apparent as noise gate (NG) output is turned on and off. Anybody can use a noise gate, but the outcome is enhanced if that person has trained ears.

noise generator • A broadband module that produces broadband random (aperiodic) signals such as white noise and pink noise. Such noise signals contain partials at all audio frequencies, with random fluctuations in amplitude and phase per each partial. See noise, white. See noise, pink. • Noise can be used to produce rushing sounds like surf (pink) or escaping steam (white). Or, heavily low pass filtered noise could be mixed with a slow periodic control signal to create "deviations" from periodicity, for example. Noise might also be used as a timing signal, particularly "Brownian" ($1/f^2$) noise or white/pink noise that has been heavily low pass filtered in order to create a slow-moving random signal.

nomenclature • A set of names and terms in a particular science or art used to differentiate items or objects. • The specific words, symbols, graphics, and numbers that name operations, functions, parameters, or features on a panel, screen, or other graphic representation in a sound synthesis system. Nomenclature can have a powerful influence on determining whether a system is easy to learn and use intuitively, or causes one gigantic, monstrous pain in the, uh, "neck."

nominal • Relating to naming, identifying, or implying "in name only." • In a sound synthesis context, a nominal bias, constant, number, or parameter value represents the initial setting, ordinary starting condition, beginning value, or starting point of a programmable parameter that may also be controlled by applying external control signals. For instance, a cutoff filter's nominal cutoff frequency is set using an associated bias (analog system) or by enumerating a constant in a virtual (digital) system. However, the actual cutoff frequency of the filter may vary widely from moment to moment relative to this nominal setting in response to the polarities (±) and amplitudes of control signals connected to that filter's external cutoff frequency control inputs. • The term "nominal" is used to describe settings determined by the user. "Default" settings are those determined by the system designer. Both terms define conditions or values that are typically static, i.e. fixed values that can be set, by the user or the designer respectively to identify (name or enumerate) an initial condition that is expected to be altered subsequently relative to this nominal/default value. Cf. default.

nonlinear device • Module or apparatus that produces, or is capable of outputting partials having frequencies not present at its signal input(s), e.g. "ring" modulator, guitar fuzz box. • Even devices that are nominally linear, e.g. amplifiers, filters, etc. can be overdriven by a signal whose characteristic(s) exceed some design specification, and the resulting distortion artifacts are typically partials with frequencies not present at the signal input. That is, even linear devices may behave in a nonlinear manner when their signal handling limits are exceeded in some way, e.g. due to exceeding input signal amplitude specifications.

normaled input • A jack with a hardwired connection of that input to a specific signal source (output) in a system. But this connection can be overridden, or changed by plugging a cable into the normaled input. This breaks the normaled input's connection with its source, i.e. the output to which it is "normally" connected. Any "substitute" signal on the cable physically connected to the normaled jack's input will still go to the destination to which the normaled jack is wired. A normaled input provides the connection most often needed between a particular signal source and the destination of the normaled input, but allows the user to break this connection in order to choose an alternative source, i.e. a different output. The destination of the normaled input remains unchanged. See default. See nominal. See Young Frankenstein (1974), Directed by Mel Brooks, for a cogent discussion of an "Abby Normal" brain. Cf. normaled output.

normaled output • A jack with a hardwired connection of that output to a specific destination (input) in a system. But this connection can be overridden, or changed by plugging a cable into that normaled output. This breaks the normaled output's connection with its destination, i.e. the input to which it is "normally" connected. A normaled output provides the connection most often needed between a particular signal's destination and the source of the normaled output, but allows the user to break this connection in order to choose an alternative destination, i.e. a different input. The source of the normaled output remains unchanged. See default. See nominal. See Young Frankenstein (1974), Directed by Mel Brooks, for a cogent discussion of an "Abby Normal" brain. Cf. normaled input.

number, counting • One of a subset of integers, which are rational numbers. Counting numbers are positive only, and do not include zero (1, 2, 3, 4, . . . 8, 9, 10, etc.). Also known as a "natural" number.

(number) integer • A rational number within the set of the real numbers, but not a fraction. Integers (-4, -3, -2, -1, 0, 1, 2, 3, . . . 4, 5, etc.) include whole numbers (0, 1, 2, 3 . . . 9, 10, 11, etc.) and natural, or counting numbers (1, 2, 3, 4, etc.)

number, imaginary • A number whose square is -1. A number that is not a real number. Cf. number, real.

number, irrational • An irrational number, a member of the set of real numbers, cannot be expressed as a ratio of natural, or counting numbers, and has an infinite non-repeating expansion when expressed as a decimal number. For example, pi ($\pi \approx 3.14159 \ldots$), and the square root of two ($\sqrt{2} \approx 1.414214 \ldots$), as well as the base "e" for natural logarithms ($e \approx 2.71828\ldots$), are irrational numbers. Any interval in the equitempered 12 tone, or dodecaphonic scale, where the frequency ratio of a half step is the twelfth root of two ($2^{1/12} \approx 1.05946 \ldots$), which is approximately a 6% factor per half step (semitone), is also an irrational number. In linear FM synthesis, a C:M (carrier to modulator frequency) ratio that is irrational yields a potential spectrum, or set of partials (sidebands) produced when full scale modulation actually occurs, that will likely include various partials not harmonic to each other, i.e. inharmonics. See ratio. See (partial) inharmonic. Cf. number, rational.

number, natural • A "counting" number that includes integers exclusive of zero and negative numbers.

number, rational • A number that can be expressed as a ratio of integral (whole) numbers. A rational number is a real number, and includes the sets of natural, whole, and integer numbers, and any ratio comprising such rational numbers, e.g. 1:5 ("one to five"), which is also expressed 1/5. Therefore, the decimal equivalent (0.20) of 1/5 is also a rational number. The partials in any harmonic series, comprising counting (whole) number ratios, are all rational numbers. In terms of musical temperament, the ratios of these partials in a harmonic series create "just intonation" intervals. • In linear FM synthesis, a rational C:M ratio (1:5, 1:2, 7:2, etc.) has a potential spectrum (group of partials produced at full modulation) capable of yielding only harmonics. That is, all the partials produced in such a spectrum will be harmonic to each other. See ratio. See (partial) harmonic. See number, real. See number, natural. See number, whole. See number, integer. Cf. number, irrational.

number, real • A point on the real number line, which includes rational and irrational numbers. See number, rational. See number, irrational.

number base • The set of unique symbols that comprise a number, or counting system is called that system's base, or radix. The base ten number system, aka "decimal," has ten unique counting symbols 0–9. Modern number systems use symbols arrayed using positional notation. Each position in a group of digits that express a quantity stands for a specific power of the base, or radix used. In the decimal system (base ten), we therefore have units (10^0 power), tens (10^1 power), hundreds (10^2 power, aka "ten squared"), thousands (10^3 power), etc. progressing from right to left from, what we refer to, due to our almost exclusive use of base ten (10) math, the "decimal" point. There are corresponding (reciprocal) negative decimal powers for tenths (10^{-1}), hundredths (10^{-2}), thousandths (10^{-3}), etc. progressing from left to right, starting to the immediate right of the decimal point. See relationship, reciprocal. Each positional number system (binary, hexadecimal, etc.) features powers of its unique base similarly. Binary (base two), octal (base eight), and hexadecimal (base sixteen) number systems (all ultimately based on powers of two) are used widely in computer systems. The duodecimal (base twelve) system is used by some scientists. • The decimal (base ten) system is commonplace, probably because we have ten fingers for counting. A symbol in a counting system is descriptively called a "digit," another word for finger, and may include both numerals and letters in modern systems. For example, the hexadecimal (base sixteen) number system uses the numerals 0–9 followed by the English letters A–F for digits, a total of sixteen different symbols. See (number base) etc.

(number base) binary • Binary (base two) is a positional number system with two symbols to express numbers, the numerals 0 (zero) and 1.

(number base) decimal • Decimal (base ten) is a positional number system with ten symbols to express numbers, the numerals 0 (zero) through 9.

(number base) duodecimal • Duodecimal (base twelve) is a positional number system with twelve symbols to express numbers, 0 (zero) through 9, and (English letters) A and B.

(number base) hexadecimal • Hexadecimal (base sixteen) is a positional number system with sixteen symbols to express numbers, 0 (zero) through 9, and (English letters) A through F.

(number base) octal • Octal (base eight) a positional number system with eight symbols to express numbers, 0 (zero) through 7.

numerical • Relating to numbers or using symbols that represent numbers. • A virtual representation of a potentiometer (pot), fader, bias, constant, or parameter value that depicts a number that can be changed, typically by click-dragging a computer mouse up for increases, or down for decreases. In some cases, a numerical allows direct entry of a number (by typing on the computer keyboard. Numericals are typically enclosed within a rectangular box on virtual modules represented on a video display terminal (VDT). Numerical(s) are used in virtual systems to program value(s) of parameter(s) as well as constants (biases). See virtual. See pot. See fader. See attenuator. See parameter.

Nyquist • Frequency that is one-half (½) the sample rate (SR) used for a pulse code modulation (PCM) sampling system. See frequency, Nyquist. The Nyquist frequency is the highest frequency that can be sampled without creating aliases, artifacts that result from an improper sampling procedure that constitute distortion of the original signal. See alias. See aliasing. See sampling theorem.

objective • (adjective) Based on information, data, or principles derived from measurements, mathematical proofs, or scientific experiments that have been replicated, and demonstrated to have apparently universal validity. Having an existence independent from any individual's thought or perception. Objective data are not necessarily unchanging "facts," but do owe their existence to inquiries not dependent on personal beliefs, opinions, thoughts, biases, feelings, superstition, guesswork, dogma, or the vagaries of subjective human perception. Cf. subjective. • That said, it should be pointed out that the important considerations about music and sound are indeed subjective, and often may appear to be at odds with objective data gathered via measurements. For example, pitch perception is a subjective evaluation that depends primarily on the frequency of a periodic audio signal or musical tone. That is, in the case of signals in the audible window (20 Hz – 20 kHz), signal frequency and perceived pitch are correlates, having direct mutual correspondences. See correlate. Frequency can be measured objectively by a frequency counter and reported in hertz (Hz), the accepted unit of measurement for frequency. See (measurement unit) hertz. However, pitch perception is also influenced by other factors such as sound intensity, waveform, duration, relative frequency range, etc. Apparent lack of agreement between subjective judgments and objective measurements usually arise not so much from real contradictions between subjective and objective worlds, but more due to attempts at drawing facile one-to-one correspondences, or simplistic correlations between these two worlds. See correlation.

offset • Equivalent expression, in terms of virtual systems, of the older analog verb "bias," an action taken using a bias (step) signal to set the initial, or nominal operating point of some parameter on a module, as in "please offset the cutoff frequency of the filter to 1200 Hz." To offset is essentially the operational equivalent of "to bias." • The authors refer to the noun form of the idea of offset as a constant, as in "the constant for the cutoff frequency of the filter has been set (rather than 'offset') to 1200 Hz." See constant.

ohm • The SI derived unit (symbol: Ω) of electrical resistance (R), as defined by the German physicist Georg Simon Ohm's (1787–1854) electrical law: $I = E / R$ where "I" is current in amperes, "E" is voltage in volts, and "R" is resistance in ohms. By rearranging this formula, resistance (R) in ohms becomes voltage (E) divided by current (I), that is: $R = E / I$. • For example, headphones and speakers are rated in ohms (e.g. 8 ohms), representing the equivalent average DC resistance, or "load" caused by the impedance, or opposition to flow of alternating current (AC), of such a transducer. The actual resistance of a transducer from moment to moment depends on several factors, including the frequencies and signal level being transduced. See nominal. See transducer. See impedance.

omnidirectional • The prefix omni from Latin omnis means "all" or "of all things." Omnidirectional means equal in directionality or sensitivity to sound waves (e.g. loudspeaker, microphone); or able to propagate or receive a signal (e.g. radio) equally in all axes, i.e. to or from all directions of three dimensional (x,y,z) space. • An omnidirectional microphone (mic) is sensitive to sound that originates from all points in a room or other environment. In contrast, a directional mic is typically more sensitive along the axis in which it is pointed.

oral • Relating to the mouth and other human anatomy used to articulate sound or speech, as in an oral contract. An oral contract is spoken, rather than written. However, both oral and written contracts are "verbal" contracts, as they both use words, rather than mime, croaking frogs, exploding firecrackers, striptease, gymnastics, etc. to communicate. (It is unfortunate that lesser dictionaries no longer make the distinction between oral and verbal crystal clear. This is another blow to communicators who value the verbal means to make such fine distinctions. "Common usage," fap! We need some uncommon lexicographers who will stand up to the riffraff.) Cf. verbal. See aural.

ordinate • The axis for "y," or the vertical value of a point on a two dimensional system of (x,y) Cartesian coordinates. See Cartesian coordinate. Specifically, the distance from a given point located on a two dimensional graph to the horizontal (x) axis, measured parallel to the vertical (y) axis. The “y” value of the ordinate is positive (+) when the point is above the central horizontal (x) axis, or negative (–) when the point is below the central horizontal (x) axis. The ordinate (y value) and abscissa (x value) form a coordinate pair, shown in the order (x,y), used to locate any point in planar Cartesian space, two dimensions graphed on a single plane. Cf. abscissa. • The (y) axis is properly known as the axis of ordinates, not simply as “the ordinate,” as there are a theoretically infinite number of distances from the (x) axis, measured parallel to the (y) axis, and therefore an infinite number of ordinate values.

oscillator • Historically, a periodic waveform generator designed to produce only a sine waveform. See waveform, periodic. The meaning of the term “oscillator” has evolved to become equivalent to periodic function generator, originally an analog periodic waveform generator that produces several geometric waveforms essentially simultaneously, e.g. sine, pulse (square and rectangular), sawtooth, triangle, i.e. geometric waveforms. See waveform, geometric. Presently, any periodic geometric waveform generator, whether analog, digital, or virtual, might be known as a function generator, or possibly as an oscillator.

oscilloscope • An electronic device or virtual system that continuously and dynamically displays one or more waveforms or other signals in the time domain. See domain, time. • Sound propagates as a longitudinal wave. However, an oscilloscope displays sound as a transverse wave, with time on the horizontal (x) axis, and amplitude, typically in terms of voltage, on the vertical (y) axis. Electronic display of a longitudinal wave (e.g. sound) would be difficult to accomplish if it were not “translated” into a transverse wave, and it would be even harder to interpret readily. See wave, longitudinal. See wave, transverse.

output • Any port, jack, or connecting device to which a cable, patch cord, or other device used to transmit signal(s) may be connected in order to route signal(s) out of a module or device. Cf. input.

output, normaled • See normaled output. Cf. normaled input.

outputs, differential • Dual outputs that provide both normal and inverted versions of the output signal, i.e. both polarities, (±) plus and minus. See jacks, differential output. Cf. jacks, differential input.

overdubbing • Recording of additional audio, particularly tracks added to a recording after other tracks have been recorded or generated previously. • In a milieu where one performer-producer may use digital audio and computer digital audio sequencer(s) to create compositions or productions one track at a time, the term “overdubbing” may become somewhat vestigial (no longer having its original meaning or utility). In some sense, most tracks in a composition created by a single person using computer sequencers and digital audio are played or programmed individually, i.e. overdubbed. The term is also taken to mean “taking the place” of an existing track.

overflow • Condition in a computer in which the maximum number, or data rate that can be accommodated due to system limitations, is exceeded. In context of audio engineering, audio distortion may result, (or at least, all heck breaks loose).

overtone • An outmoded term used to identify partials above the fundamental, the first "overtone" being the second harmonic (H2), and so forth, in typical (mis)usage. In recommended modern usage, the fundamental is the same as the first harmonic (H1), where "harmonic" is defined as an integer, or counting number (1, 2, 3, etc.) multiple of the fundamental, numbered consecutively starting from the first harmonic (H1), the fundamental. That is, the first harmonic (H1) is one (1.0) times the fundamental–it is the fundamental. And, the 2nd harmonic (H2) is a doubling (2.0 times) the fundamental frequency (H1), even when other partials (known as inharmonics, or nonharmonics) lie between the fundamental (H1) and the 2nd harmonic (H2). The utility of the confusing term "overtone" is doubtful, and its use is discouraged. The distinguished physicist and sound researcher Hermann von Helmholtz inveighed against its use during the 19th century. The preferred term partial is used to identify any individual sine (or cosine) wave that is part of the spectrum of a complex waveform, whether that partial's frequency has an harmonic (integer) or inharmonic (non-integer) frequency ratio to the fundamental frequency (H1). All harmonics and inharmonics are partials. But, not all partials are harmonics. That is, a partial may be either harmonic or inharmonic, determined by its mathematical frequency relationship, or frequency ratio with the fundamental (H1). The term "overtone" has no utility in such a straightforward context. See partial. See (partial) harmonic. See (partial) inharmonic.

oversampling • Data conversion technique by which each sample in an existing digital audio file is replicated, or duplicated a number of times. The "new" samples are typically processed to create interpolated (intermediate) values that lie between the original sampled values. The purpose is to increase the Nyquist Frequency of the sound file, effectively yielding an apparently higher sample rate (SR). A higher sample rate facilitates system design featuring an output "brickwall" (smoothing) filter with a less severe slope, with attendant reduction of phase anomalies in the final (analog) output of the sampling system. With oversampling it becomes possible to effect "smoothing" in the digital domain, thereby relieving the severity of design constraints on the analog "smoothing filter" that reconstructs the original analog signal from its digital counterpart. See filter, smoothing. The extent of oversampling is often expressed by a coefficient, e.g. "2X, 4X, 8X, 16X" indicating the factor, e.g. "2" by which the original number of samples "X" is multiplied. For instance, a compact disk (CD) with 44 100 samples per second, will have 88 200 samples per second after being oversampled at "2X." This oversampled version of the sound file will have a Nyquist frequency (44 100) that is twice (2X) that of the original (22 050). See frequency, Nyquist. This increase of Nyquist frequency does not affect the pitch level of the sound file that is oversampled.

overwrite • To cause data to be stored, or written to computer memory (of any type), thereby destroying existing data in the computer addresses used, i.e. memory locations used to store the new data. Existing data in memory are lost when overwritten with new data.

oxymoron • An oxymoron is an expression of speech that includes words or roots that have mutually contradictory meanings, e.g. “virtual reality.” See virtual reality. The term “negative gain” also comes to mind, although this usage is accepted, and is technically correct. See amplifier, negative gain. The word oxymoron has roots oxy meaning sharp, and moros meaning dull, from which we derive the word moron–moron. In general, a person who mindlessly propagates, perpetuates, or even tolerates oxymorons (and highly educated morons) often exhibits a personality profile that mirrors both roots in the word: a moronic ox. With regards to the oxymoron “virtual reality” search also “media hype, group think,” and “horse foofy” (these last three terms not included in this Learner’s Glossary). Uh, “group think” is certainly an oxymoron, come to think of it–no group needed, and committees usually keep minutes and waste hours. Horse foofy and media hype are both real, and palpably unavoidable during parades, but at least horse foofy is good for something.

pad • Switchable negative gain (attenuation) stage on a recording console or other device that, when engaged or enabled, attenuates (reduces) an input signal by a fixed power ratio, e.g. –20 dB. The 20 dB loss in this example represents a 1:100 power ratio, which is equivalent to a 1:10 signal amplitude or voltage ratio. (The disparity between ratios here is due to the square relationship between signal amplitude. e.g. as measured in voltage, and power, as measured in dB (decibels) or some measurement of power such as the Watt.) See relationship, square. See (measurement unit) decibel. • A pad is typically used to reduce a larger line level, e.g. a synthesizer output signal, to a lower level that more closely approximates microphone (mic) or electric guitar output level.

panning • Moving the apparent location of an audio signal played by more than one speaker, by setting a pan pot (potentiometer). See pan pot. • In film or television, to turn a camera in a horizontal plane in order to keep a moving object in view, or to show a panoramic effect or view of the locale.

pan pot • A (rotary) control on each “strip,” or channel of a recording console used to position that channel’s audio signal in a specific location in the stereophonic listening field. A pan pot allocates different levels of the same signal between left (L) and right (R) speakers in a stereo configuration. In stereo, as a pan pot is turned, signal level is increased in one speaker, and proportionately decreased in the other speaker. The apparent location of the sound is “fixed” between speakers based on the current location of the pan pot. The effect is not dynamic–it is static, i.e. determined by the pan pot’s current position. This sound location capability is known as panning. • With the advent of viable surround sound configurations (e.g. 5.1 surround, 7.1, etc.) the “panoramic” sound field is extended into putatively three dimensional space, rather than nominally two dimensional L–R stereophonic space. Also, panning may be “automated” in some systems, such that pan pot position may be moved dynamically by a selected control signal, particularly one that has “remembered” prior knob movements and stored those data in computer memory.

parameter • A measurable quantity of an individual element located within a module or other device in a sound synthesis system, with a signal value, setting, or number that can be changed, or programmed and stored in some form of memory. Oscillator waveform, filter cutoff frequency, envelope attack time, etc. are examples of parameters. • In DSP terms, a parameter is a variable or a constant programmed with a specific value to allow successful execution of an algorithm, set of encoded instructions, or computer software (e.g. a virtual sound synthesis module). • The word parameter is borrowed from mathematics and statistics, where the term is defined more strictly than its usage in modular sound synthesis implies.

partial • The simplest and singular component of the spectrum of a complex waveform, a sine (or cosine) waveform, having a specified frequency, amplitude, and (in some cases) phase. A partial is part of, i.e. an individual element of a complex waveform. A complex waveform may include partials that are harmonic and/or inharmonic to its other partials. Two sine waves (partials) having frequencies that comprise an integral ratio, or a ratio that resolves to whole numbers (e.g. 0.5:1 which resolves to the integral ratio 1:2), i.e. a rational number, are harmonic to each other. See number, rational. All of the musical intervals possible among members of any harmonic series constitute ratios that are rational numbers, and therefore all members of a harmonic series are harmonic to each other, i.e. they are harmonics. See (partial) harmonic. When the frequency ratio of two sine waves (partials) is not a rational number, those partials are inharmonic to each other. See (partial) inharmonic. That is, paraphrasing a dictum from symbolic logic class: “all harmonics are partials, but not all partials are harmonics.” Also, “all inharmonics are partials, but not all partials are inharmonics.” • Uh, for those of us who slept through those symbolic logic classes: all horses are quadrupeds, but not all quadrupeds are horses. Yes. Thinking logically can be quite useful, especially when dealing with your Business Manager. When thinking logically, it is also wise to simplify, as per the clarion edict: eschew obfuscation to obviate imbroglios! While you’re at it, avoid the outmoded term “overtone” like the plague (oops, we accidentally used the dreaded “O” word just now–quite the story). Simplify, Mr. Satie! Use only the term partial to refer to: (1) any harmonic, including the first harmonic (H1), aka the fundamental of a periodic waveform; (2) any inharmonic; and (3) any personal dental appliance that facilitates chewing, rather than eschewing.

(partial) harmonic • Sine wave whose frequency has integral (whole number) relationships with the frequencies of other harmonics in a set of partials that constitute a complex waveform, and/or with all of the frequencies in a harmonic series. See harmonic series. The frequency ratio of one harmonic to another is a rational number, i.e. one that resolves to an integral ratio comprising only whole numbers, such as 3:5 (the interval of a Major 6th (M6), a “just temperament” Major 6th). An “equal temperament” Major 6th does not resolve to such an integral ratio. See temperament. Each harmonic has a frequency that is a whole number multiple (1, 2, 3, 4, 5, etc.) of the frequency of the fundamental, or first harmonic (H1) in a periodic complex waveform. The fifth harmonic H5 is five (5) times the frequency of the first harmonic H1, and so forth. The fundamental (1/1) of a harmonic series is the first harmonic (H1), notwithstanding alternative application of the term “harmonic” by guitarists that excludes the fundamental, which actually is the first harmonic (H1) in acoustics or physics. See waveform, periodic. See waveform, complex. Cf. (partial) inharmonic.

(partial) inharmonic • Sine wave whose frequency does not have an integral (whole number) relationship with the first H1, or various other harmonics in a given set of partials that comprise a spectrum. An inharmonic, aka as a "nonharmonic," has a frequency that is not a whole number multiple of the fundamental frequency (H1) of a complex waveform. • Inharmonics do not necessarily reinforce the frequency of the first harmonic H1, or fundamental, i.e. the pitch we perceive. A bell has a spectrum that includes inharmonics, which may cause the bell's pitch to be judged differently by listeners. Bell sounds are clangorous, implying a discordant or "clang" sound, unlike a bowed violin tone, all of whose partials are very nearly exact harmonics. See clangorous. See waveform, complex. See spectrum, line. Cf. (partial) harmonic. • The term inharmonic should not be confused with the term enharmonic. See enharmonic.

pass band • See band, pass.

patch • Configuration of a specific interconnection of modules intended to generate and/or process signals of interest. A modular sound synthesis patch typically comprises connections among modules designed to create a particular sound. A patch diagram comprises icons that represent modules, with lines indicating interconnections.

patch bay • An array or matrix of jacks that provides centralized access to the inputs and outputs of various devices, modules, processors, and/or rack-mounted equipment in a recording studio. See jack/plug.

patch cord • Wire or cable with appropriate plugs affixed to both ends, that may be inserted into jacks on modules or patch bays in recording studio(s). Plugs are "male," and jacks are "female." (Hey, don't look at me, I didn't make up this sexist language.) • A patch cord may be a virtual representation of such a connecting device, which has no need to represent particular kinds of physical plugs or jacks. See jack. See plug.

period • The amount of time (T) a periodic signal, wave, or other phenomenon takes to complete its waveform, cycle, or occurrence. Period (T) is the reciprocal (inverse) of frequency (f), that is: ($T = 1 / f$); and frequency is the reciprocal of period, that is: ($f = 1 / T$). Cf. frequency. Audible periodic vibrations within the frequency span of human hearing (20 Hz – 20kHz) have a perceived "pitch." See pitch. See relationship, inverse. See periodic.

periodic • Signal, wave, or other phenomenon that completes its cycle or occurrence during a specific interval of time (T), and typically repeats that cycle indefinitely during identical ensuing time intervals. A periodic wave is represented by repeated transitions from 0° (zero degrees) through 360° of its waveform during its ensuing period(s) (T). See phase. See period. See frequency.

phase • A temporal (time) consideration or time anomaly represented by a transition through a specific number of degrees of arc where one complete circular motion is 360 degrees (360 °). • In wave motion, the amount of time that a fraction (less than 360 ° phase relationship) or a multiple of a wave's period (greater than 360 ° phase relationship) takes after last passing through its zero position or some other reference point on the wave(s). • When two waves of the same frequency are in phase their initial zero positions are aligned in time, and their interference, or algebraic summation is constructive. Two waves of the same frequency are 180° (degrees) out of phase when the initial zero (0 °) point of one is aligned in time with the 180 ° point of the other, and their interference, or algebraic summation in this case is destructive. See interference, constructive. See interference, destructive. Cf. polarity.

phaser • Processor that includes one or more all pass filters, each of which causes a phase change relative to the unprocessed signal, ideally without altering the amplitude of any partial processed. The output of each all pass filter is summed with the "dry," or unprocessed signal, creating as many peaks or valleys, i.e. boosted or attenuated frequency bands, as there are all pass filters. These peaks and valleys, which may begin to approximate the frequency response curve of a comb filter, are typically moved dynamically using an internal low frequency oscillator (LFO) that changes the center frequencies, i.e. the timing constants of the all pass filters. See filter, all pass. Cf. filter, comb. • A phaser is not to be confused with a flanger, which can sound similar to a phaser. Flanger operation depends on adding a single delayed version of the processed signal to the original signal, and changing this single delay time dynamically using a low frequency oscillator (LFO). See flanger.

phi (Φ) • (pronounced like "fie"). The 21st letter of the Greek alphabet (Φ), often used to represent the "golden section," a ratiometric division of a line, or the ratio of two dimensions or elements found in classical architecture, art, music, etc. The phi, or golden section ratio is deemed by many to be pleasing to the eye (and possibly, the ear as well). This ratio, approximately 1.618:1 is derived from the general formula $(\sqrt{5} + 1)/2$. If the relevance of Φ is not apparent, just look at many of the icons that represent modules, and other splendid graphics in this publication. If you still don't understand, then Φ on you!

pitch • Subjective sensation due to perception of an audible periodic musical tone. Pitch provides a sense of relative highness or lowness of a tone with respect to the gamut, or complete span of possible pitched musical notes, from low to high. Pitch perception is subjective, but correlates strongly (and positively) with the frequency of a periodic wave, which can be measured in hertz (Hz), formerly cycles per second (cps). Frequency is an objective signal characteristic that can be measured. Pitch is the subjective evaluation of a sound that has a frequency, specifically the perception of sound produced by a periodic vibration in the range of human hearing. • Audio signal intensity is perceived as loudness when all other factors are held constant (not changed), and loudness plays a secondary role in our evaluation of pitch. Timbre (tone color), duration, envelope (transient times), etc. also play subtle roles in pitch perception. That is, our sense of pitch does not "map," or correspond to, i.e. correlate with the objective frequency of a tone perfectly. Many factors affect our sense of pitch. The implication of the known non-linearity of frequency-pitch correlates is that musical instruments and composition structures should be tuned by an informed ear, not via mechanical or electronic means. • The most popular tuning standard for pitch in a musical key (tonal center) dictates that the frequency of the note A4 (the note A above middle C on the piano) is tuned to 440 Hz, with frequencies of other notes in the scale being tuned proportionately in some temperament system. See temperament. This frequency (440 Hz) is aka "tuning A." Cf. frequency. See aural. See subjective. Cf. objective. See correlate. See transients, attack. See transients, decay.

pitch to frequency converter • Interface device or system that senses a musical pitch (note) being played or sung, and outputs a specific value, e.g. the MIDI note number that corresponds approximately in frequency to that pitch. This conversion is burdened with problems–and is consequently only approximate, as pitch and frequency do not correlate perfectly. See pitch. See frequency. See correlate.

plug • The "male" part of a cable or electronic signal connection system that may be inserted into a jack, its "female" counterpart. See jack/plug connectors. Cf. jack.

plug/jack connectors • See jack/plug connectors, etc.

polarity • The direction of an electrical flow or orientation of a magnetic field, expressed as either positive (+) or negative (–) with respect to zero (0).

(polarity) bipolar • A signal or its virtual representation that includes both positive (+) and negative (–) levels or numbers over time. That is, a bipolar signal does not have both positive and negative levels at the same time. • A waveform or signal that crosses zero (0) level in order to exhibit both plus and minus (±) polarities or numbers. Alternating current (AC), or common household electrical power, is an example of a bipolar waveform. Cf. (polarity) unipolar.

(polarity) unipolar • A signal or DSP representation that includes either, but not both positive (+) and negative (–) levels or numbers. A wave or constant may be either unipolar positive or unipolar negative. • A waveform or signal that typically includes 0 (zero), but does not cross zero level to become bipolar. See (polarity) bipolar. Direct current (DC), e.g. a "battery" of cells, of either polarity (+ or –) is an example of a unipolar waveform. Cf. (polarity) bipolar.

pole • A section of filter circuitry that causes attenuation of the input signal, with a slope of –6 dB/Octave, i.e. a 6 dB loss per each successive octave progressing into that slope's associated stop band. See band, stop. The effect of concatenating, or connecting several poles in series is additive. That is, each additional pole accounts for an additional 6 dB of loss (– 6 dB/Oct), creating a progressively steeper filter slope. For example, a four pole filter has a filter slope between pass band and stop band of –24 dB/Octave. See filter slope. See concatenate.

polyphonic • Musical instrument capable of producing two or more musical voices or lines simultaneously. Cf. monophonic. • Literally "many voices," indicating music that features lines that can change pitch or move independently. See (music) polyphony.

port • General term for an input or an output. In terms of modules, a port is a terminal– it stands on the "frontier" of that module. See module.

portamento • Continuous, or gliding changes of pitch during the transition from one musical tone to another. The term portamento is associated with continuous pitch transitions, particularly those made by the singing voice, rather than the discrete changes produced by acoustic keyboards. On electronic music synthesizers, portamento-like changes of pitch can be produced by enabling "glide." See glide. • Portamento has been confused by some with glissando, a performance of discrete notes serially from a starting to an ending note, usually done by rapidly strumming or "glissing across" keys on a keyboard (or harp strings) with fingers, or the nails of fingers. Cf. glissando. See glide.

post production • In terms of sonic arts, any replacement, addition or alteration (aka "sweetening") of recorded sound following some prior preparation or action, such as filming. Most dialogue, foley, and sound effects (SFX) for film and video productions are produced "post" (after film is exposed, or video is shot). Dialogue is often added using techniques such as automated dialogue replacement (ADR), formerly called looping. In a complete post production facility, sounds created in real time by foley artists, e.g. footsteps, body, and clothing sounds, are also recorded as the foley artist(s) synchronize their movements to actions of actor(s) pictured, or other visual elements previously recorded. See foley. See SFX.

pot • Abbreviation for potentiometer. See potentiometer.

potentiometer • Abbreviated pot. An electronic hardware processor that comprises a variable resistor that divides the input signal into two paths in order to attenuate, or reduce the signal. One of these paths passes to the output of the pot, the other path is shunted to electrical ground through a resistor, a configuration that allows part of the energy of the processed signal to be dissipated as heat, causing signal attenuation. • In music synthesizers, samplers, and audio engineering hardware, a pot acts as an interface between human and electrical elements of the system. Attenuators on the audio recording console may be slide-pots, as represented by ubiquitous (seen everywhere–like Elvis) console faders; or rotary (circular) pots, typically used for equalization (EQ) controls. Most hardware potentiometers are attenuators that directly reduce signal amplitude. In automated audio engineering consoles, a fader or other pot may provide a signal that controls associated VCA or SCA gain, thereby indirectly providing attenuation. See attenuator. • In a virtual environment, the function of a pot is typically effected mathematically using division, aka attenuation (reduction) of a signal. An attenuator may be depicted as a slider, rotary control, or a rectangular graphic box in which numbers appear that can be altered, often by computer mouse movements. See numerical.

precedence effect • Phenomenon relating to the perception of a sound's location, particularly in a stereo audio configuration, based on the speaker from which a sound is first produced, rather than due to its panoramic position based on relative sound intensities in left (L) and right (R) speakers. That is, a sound that is actually stronger in left (L) might appear to be located more toward right (R) if that sound comes from right (R) first, and then appears in both (L-R) soon thereafter. This is also related to the Haas Effect, and is most effective when the two sounds appear with no more than 30–40 milliseconds difference in time, and the difference in their levels is no greater than 10 dB.

pressure • MIDI after touch has erroneously been called "pressure," but pressure is a measurement of force exerted per unit area (p = F/s where "s" is unit, or "surface" area). Air pressure @ 30 pounds per square inch (psi) in automobile tires is an appropriate example of pressure. Cf. force. MIDI after touch is an example of force. See after touch.

pressure compression(s) • Location or time of highest pressure(s) of a sound wave as it propagates in a fluid, e.g. air without interference. See interference, etc.

pressure rarefaction(s) • Location or time of lowest pressure(s) in a sound wave as it propagates in a fluid, e.g. air without interference. See interference, etc.

processor • A "two-ended" device having both a signal input and a signal output, capable of altering the signal that passes through it. • See (module) processor.

product • The mathematical result of the multiplication of two or more factors that are numbers or quantities. See factor.

proportional • Description of a quantitative relationship where two ratios are equivalent. A variable quantity that maintains a constant ratio to another quantity. Ratios 5:10 and 100:200 are proportional because 5:10 = 100:200 (each ratio represents a halving, and 1:2 = 1:2 indicates the proportionality of these two ratios). See ratio. • A given musical interval maintains proportional changes of frequency at any point on the musical scale. For example, the frequency ratios of an octave in music remain the same in any pitch range, e.g. 440 Hz : 880 Hz (octave from note A4 to note A5), or 220 Hz : 440 Hz (octave from note A3 to note A4). That is, 440 : 880 = 220 : 440 (each ratio is a halving of frequency (1:2) or doubling (2:1), based on the note at which one starts; therefore, the two ratios are proportional). See ratio. See relationship, etc.

Q • See filter Q. See resonance.

quantization • Process that yields discrete approximations of analog signal levels that are originally free to fluctuate using continuous signal levels or numerical values whose resolutions is theoretically unrestricted. Quantization produces a signal limited to discrete steps, or samples, each of which having a possible numerical value constrained by a limited set based on the number of bits (binary digits) used in the quantizing process. Quantization produces quantizing errors (quantization noise), but these errors can be made vanishingly small or practically negligible by increasing system resolution (number of bits used when quantizing). See noise, quantization. See (measurement) resolution. • Quantizing, or quantization error is sometimes confused with the bandwidth limitations that accompany a selected sample rate (SR). The number of bits used in sampling determines the amount, or relative power of quantization noise created when sampling, not the available bandwidth (frequency limitation). Sample rate (SR), rather than the number of bits used when quantizing, determines the total bandwidth of frequencies that can be sampled successfully without producing aliases, or partials with "counterfeit" frequencies. Quantizing error (involving the number of bits used) is properly referred to as "noise," while aliases caused by violations of the Nyquist frequency, a sampling rate consideration, might best be called "distortion." This distinction is not always adhered to throughout the literature. See alias. See aliasing. See sampling theorem. See frequency, Nyquist.

quantization noise • Aka quantizing noise. See noise, quantization. See quantization.

qwerty principle • The contiguous keys q-w-e-r-t-y (spoken "kwerty" as in a riding quirt) on a typical ASCII keyboard are used to signify a particular layout (location) of all keys on such a keyboard. See ASCII. The qwerty principle refers to an idea in the history of ergonomics (human engineering, man-machine interface considerations, or biotechnology) that the "first" solution to a problem may not be the "best," even though the first solution tends to persist, often maintaining a presence in the marketplace forever. The peculiar (qwerty) layout of the keys on the early "standard" typewriter originated from the vestigial (outdated) need to prevent mechanical elements from jamming, during the era when the typewriter was strictly mechanical. The qwerty keyboard layout was purported to prevent such jamming, perhaps with some justification. However, within a number of years the jamming problem was solved more elegantly, but the qwerty keyboard layout persisted. Much later, research by the U.S. Navy and others confirmed that the newer and ergonomically superior Dvorak keyboard layout facilitates superior typing accuracy at a faster pace. Nevertheless, the original qwerty keyboard layout has prevailed in the marketplace, even though its original justification is vestigial (having an historical basis or original utility that is no longer necessary or valid). This is undoubtedly due to the early appearance of qwerty keyboards, and possibly their initial sole availability in the marketplace, and subsequent adoption by most typists as a de facto standard in the absence of something better. The qwerty principle remains an object lesson for designers today to "get it right the first time." (The first time may be your only opportunity–if you are sufficiently fortunate to create a de facto standard.) See ergonomics. See de facto.

radix • The number of symbols in a positional ("base and place") counting system such as decimal (radix, or base of ten with symbols 0–9). See number base.

random • Action, choice, decision, or event that occurs without recourse to a specific predetermined plan, pattern, or known mathematical sequence or known probability of the occurrence of predictable numbers. Pertaining to a mathematical set where all elements have the same probability of occurrence or another probability of occurrence based on a particular (e.g. Gaussian bell-shaped curve) distribution. • Pertaining to variables that have an unknown or undetermined value but a known probability, or likelihood of occurrence, referred to in statistics as a "distribution."

ratio • Quantitative relationship between two values that indicates the number of times one value is contained within the other. Ratio is represented by a fraction that expresses the relationship between, or comparison of two similar measures, e.g. two sound intensities (powers) in decibels (dB), lengths of two lines, e.g. that represent the "golden section" ratio phi (Φ sounds like "fie"), etc. See phi. The fraction that represents a ratio, e.g. "1/5" can also be shown as "1:5" with a colon (":") between numerator ("1") and denominator ("5"). This example (1:5) would be spoken as a "one to five" ratio. A ratio does not have to be a rational number such as the example given, which is a number expressible as a fraction using two integers (whole numbers). For example, the ratio for pi (π, pronounced like "pie"), is approximately 3.14159:1, which is not a rational number. See number, rational. • The decibel (dB) scale features a ratio of two powers, as per the formula for dB in sound intensity level: $SIL_{dB} = 10 \log (W_1 / W_0)$, e.g. where "W" is in watts, a unit of measurement for power. See (measurement unit) decibel. • The musical interval of an octave is produced by two notes with a frequency ratio of 2:1 (or 1:2), depending which of the notes is the reference for the octave interval, the lower or higher. Quite literally, one multiplies by two (2:1) or one half (1:2) to create a frequency an octave higher (2:1) or lower (1:2) than any given starting frequency. See interval, musical.

reciprocal • A reciprocal is a non-zero mathematical number or element, e.g. "x" such that "1/x" is known as the reciprocal of "x." When a number and its reciprocal are multiplied, the product (result of multiplication) equals one (1), or unity. See relationship, reciprocal.

rectifier • Electronic circuitry with diodes and associated components that changes alternating current (AC) into direct current (DC). A rectifier is typically an integral part of a DC power supply that uses AC as an electrical energy source. See diode.

relationship • Mutual correspondence, similarity, or connection between two ideas or things, or between elements in two different sets of numbers or other data.

relationship, direct • Mathematical relationship of two variables where both variables increase together, or decrease together according to some common factor or ratio. See factor. See ratio. • Period (T) and wavelength (□) of a periodic waveform have a direct relationship: as the number representing period becomes larger (or smaller), the number for wavelength becomes proportionately larger (or smaller). Wavelength is represented by the Greek letter lambda (□). Sound propagates in air at room temperature (20° Celsius) at a velocity (v) of 344 m/s (meters per second). The formula that relates velocity (v), frequency (f), and wavelength, or lambda (□) is: v = f □. Solving for the wavelength of a frequency of 1000 Hz: □ = v / f = 344/1000 shows that 1000 Hz has a wavelength, or lambda (□) = 0.344 meters–slightly larger than a "foot long" hot dog (frankfurter). By similar calculation, 500 Hz has a wavelength (□) = 0.688 meters. Note that frequency and wavelength in this example have a relationship that is not direct– it's an inverse relationship: as frequency decreases, wavelength increases. However, frequency ($f = 1 / T$) and period ($T = 1 / f$) also have an inverse relationship. Therefore, period (T) and wavelength (□) logically have a direct relationship, as stated previously. So, a waveform with a period (T) of 1 millisecond (corresponding to a frequency of 1000 Hz) has a wavelength (□) of 0.344 meters. A waveform with a period (T) of 2 milliseconds (500 Hz) has a wavelength of 0.688 meters, thus illustrating the direct relationship between period (T) and wavelength (□). As period grows larger, wavelength becomes proportionately larger, as per the 2:1 ratio (doubling) of both values in this example. Conversely, as period decreases, wavelength also does so proportionately. See proportional. Cf. relationship, inverse.

relationship, inverse • Mathematical relationship of two variables where one variable increases proportionately as the other variable decreases, and the converse. See relationship, reciprocal. Cf. relationship, direct.

relationship, monotonic • Mathematical relationship of a function or related sets of quantities. A monotonic function or relationship changes such that it either solely decreases or solely increases, i.e. the change is in only one direction. A relationship is monotonic increasing when an increase in the value of one variable is accompanied by an increase or non–decrease in the value of the other variable (a positive correlation). A relationship is monotonic decreasing when an increase in the value of one variable is accompanied by a decrease or non–increase in the value of the other variable (a negative correlation). See correlation. • A monotonic curve may exhibit either increasing or decreasing returns (directionality of curving), but not both. A monotonic relationship cannot embody a "U-shaped" curve. See relationship, U-shaped. Logarithmic ($y = \log_{10} x$) functions produce monotonic increasing curves of decreasing returns (the curve flattens out as it decreases), and exponential ($y = a^{x}$) functions produce monotonic increasing curves of increasing returns (the curve becomes steeper as it increases). • In music synthesis, a monotonic relationship exists between the progressively increasing harmonic numbers in a sawtooth wave being filtered, e.g. by a low pass filter, with the higher cutoff frequencies to which that filter is tuned. That is, progressively higher harmonics appear only when progressively higher filter cutoff frequencies are tuned. And, all harmonics below the current cutoff frequency remain present. • To give an example of a relationship that is not monotonic, the amplitudes of (sideband pairs), i.e. the harmonics produced in the case of an integral linear FM (frequency modulation) carrier to modulator (C:M) frequency ratio do not have a monotonic relationship with the index of modulation (a number that represents the depth of modulation and resulting bandwidth produced. That is, a higher index of modulation does not necessarily ensure that the same sideband pairs of harmonics (or non-harmonics) that were present at a lower index of modulation will be present at a higher index of modulation. Nor does a higher index of modulation ensure that the amplitude of a particular sideband pair will be greater. Sideband pairs of partials produced due to linear FM have amplitudes that change and sometimes completely disappear, change polarity (±) and reappear, as determined by Bessel functions, which are not themselves monotonic functions. Neither the actual presence of particular partials nor the amplitudes for those partials currently present maintain a monotonic relationship with the index of modulation in linear FM. To give another example, Robinson-Dadson (Fletcher-Munson) curves of equal loudness exemplify a set of curves that are not monotonic, they are "U-shaped." See relationship, U-shaped. See loudness curves, equal.

relationship, proportional • Mathematical relationship of two variables that maintain a constant ratio when either one of the variables is changed. That is, as one element in the ratio is changed by a given factor, the other element in the ratio must be changed by the same factor if the relationship remains proportional. For example, if the first number in the ratio 1:2 is tripled, i.e. changed to "3," then the second number must also be tripled, i.e. changed to "6," to maintain the 1:2 ratio, then expressed as 3:6. That is, 1:2 = 3:6 represents a proportional relationship. See ratio.

relationship, ratiometric • See ratio.

relationship, reciprocal • A reciprocal is a non-zero mathematical number or element, e.g. "x" such that "1/x" is known as the "reciprocal of x." The reciprocal of "x" is aka the "multiplicative inverse of x." Any given number such as "x" equals unity, or one (1) when multiplied by its reciprocal "1/x." • Frequency (f) and period (T) of a periodic waveform have a reciprocal, or inverse relationship. Frequency is the reciprocal of period ($f = 1 / T$), and period is therefore the reciprocal of frequency ($T = 1 / f$) due to this inverse relationship. For example, a waveform with a frequency (f) of 1000 Hz (cycles per second), has a period (T) of 0.001 (one thousandth) of a second, aka one (1) millisecond. The number 1000 can be expressed as 10^{3} (ten to the third power); the reciprocal of 1000 is 1/1000 (one-thousandth), which can also be expressed as 10^{-3} (ten to the negative third power). A positive (+) and negative (–) power with the same numerical value (e.g. 3) have a reciprocal, or inverse relationship. See absolute value. See number base, decimal. • In a sawtooth waveform, the amplitude of any harmonic relative to the amplitude of the fundamental, or first harmonic (H1) is the reciprocal of its harmonic number. For example, the amplitude (y) of, e.g. the first harmonic (H1) is computed ($y = 1 / x$), where "x" is the harmonic's number (e.g. 1), which equals 1/1 or more simply one (1), which equals full scale, i.e. the maximum amplitude that any harmonic can attain in our example. And, 1/3 is the amplitude of the third harmonic (H3) with respect to full scale amplitude ($y = 1 / x = 1 / 3$). That is, the amplitude of any harmonic in a sawtooth waveform has a reciprocal relationship with its harmonic number. (The frequency of any harmonic in a sawtooth waveform has a direct relationship with its harmonic number, e.g. the third harmonic (H3) is three (3) times that of the fundamental (H1). See relationship, direct).

relationship, reciprocal square • A reciprocal is a non-zero mathematical number or element, e.g. "x" such that "1/x" is known as the "reciprocal of x." To square a mathematical term or number is to multiply that term or number (n) by itself, e.g. indicated by a superscript (elevated) numeral "2" thusly: "n^2." • A reciprocal square relationship between variables "y" and "x" takes the form ($y = 1 / x^{2}$). See relationship, reciprocal. See relationship, square. • In a triangle waveform, the amplitude "y" of any harmonic is the squared reciprocal, aka reciprocal square of that harmonic's number "x." For example, the amplitude of the first harmonic (H1) computed by ($y = 1 / x^2 = 1 / 1^2$) is 1/1 or more simply one (1), which equals full scale, i.e. the maximum possible amplitude of any harmonic in this waveform example. Therefore, ($y = 1 / x^2 = 1 / 3^2$) reveals that the amplitude of the third harmonic (H3) is one ninth (1/9) that of the amplitude of the first harmonic (H1), aka as the fundamental. The amplitude of any harmonic in a triangle waveform has a reciprocal square relationship with its harmonic number. • The inverse square law is a general law in physics that describes many phenomena, e.g. sound power loss in a free (non-reflective) field, and it involves a reciprocal square relationship. In terms of acoustics, this law states that, for every doubling of distance from a point source of sound in a free field having few or no reflections, there is a halving of amplitude, so there is a quartering of power, which is a six decibel (–6 dB) loss. That is, ½ is the inverse of 2 (doubling is a 2:1 ratio, or 2/1), and quartering (¼ , or 1:4) is the result of squaring one half (½ , or 1:2). That is, one half times one half equals one fourth (½ x ½ = ¼). A 1:2 ratio of amplitude equals a 1:4 power ratio, and either represents a six decibel loss (–6 dB). The decibel represents a power ratio, and almost everything else of interest to us (amplitude, voltage, current, sound pressure level (SPL) has a square relationship with the decibel. In the case of distance, there is an inverse square relationship, due to the fact that sound power diminishes over distance. See (measurement unit) decibel.

relationship, square • To square a mathematical term or number is to multiply that term or number by itself. • A square relationship is a mathematical relationship of two variables where, as one variable "x" increases, the second variable "y" increases by the square of "x," expressed as the formula: ($y = x^2$) • For example, signal amplitude and power have a square relationship. That is, a doubling (2:1 ratio) of amplitude "x" results in a quadrupling (4:1 ratio) of power "y," shown as ($y = x^2 = 2^2 = 4$). A tripling (3:1) of amplitude "x," shown as ($y = x^2 = 3^2 = 9$), causes power "y" to increase ninefold (9:1), which is (3^2). Conversely, to solve for the amplitude change that results in a given power change, the inverse of squaring must be done, i.e. finding the square root. That is, taking the square root of "4" (a quadrupling (4:1) of power "y") reveals that a doubling "2" (2:1) of amplitude (or voltage) "x" is equivalent, shown as ($x = \sqrt{y} = \sqrt{4} = 2$); and the square root of "9" (a ninefold (9:1) increase of power "y") reveals that a tripling "3" (3:1) of amplitude "x" is equivalent, shown as ($x = \sqrt{y} = \sqrt{9} = 3$). Squaring and finding the square root are inverse math operations. See relationship, inverse.

relationship, U-shaped • Mathematical relationship or function of two variables, or of a particular set of values of a function where the value of one variable increases or decreases when the value of the other variable increases, i.e. are not in a monotonic relationship • A U-shaped curve is a graphical depiction of such a relationship or function, and is "bidirectional." • The frequency response of human hearing is an archetypal representation of an inverted U-shaped curve, with maximum sensitivity at 3.5 kHz and decreasing sensitivity as both lower and upper frequency extremes of the audible window (20 Hz–20 kHz) are approached. See loudness curves, equal. Cf. relationship, monotonic.

release • The timing and manner (curve) over which an articulated sound's intensity falls to silence. See transients, decay. • Final release (R) segment produced by an ADSR envelope generator. The release segment typically falls to zero level from some positive level over a time (R) set by the user. In some designs, envelope generator release (R) time may be subdivided into several (shorter) segments, each of which features two programmed level(s) with an associated time, facilitating changes of the curve of the overall release (R) segment. See EG segment.

residue pitch • Perceptual phenomenon that makes it possible to identify a pitch whose fundamental frequency, or first harmonic (H1) is not physically present. Residue pitch depends on the human ear's ability to analyze upper harmonics spaced at equal frequency distances for adjacent harmonics, expressed in units of hertz (Hz) equal to the fundamental frequency. The ear's analysis makes it possible to "fill in," or deduce the missing fundamental frequency, and make a valid judgment regarding pitch. Harmonics are whole number (integer) multiples of the fundamental frequency, i.e. 1x, 2x, 3x, 4x, etc. where "x" represents the fundamental. Any two adjacent harmonics, e.g. H1–H2, H12–H13, H55–H56, etc. differ from each other by the frequency of the fundamental. That is, progressive whole number multiples of the fundamental frequency (harmonics) might also be thought of as progressive additions of the fundamental to itself. These relationships are generic to any harmonic series. • To give an example of residue pitch in operation, a singer's voice does not change pitch when it is high pass filtered, even though the fundamental, and likely several other low order harmonics are "cut off," i.e. removed completely. When the ear hears upper harmonics in sufficient numbers and strengths, those harmonics that are actually physically present tend to reinforce perception of the frequency of the missing fundamental, and its "residue pitch" is clearly perceived, even though that fundamental's corresponding frequency is not actually present physically, i.e. the fundamental frequency is not actually sounding.

resonance • Control on a filter that increases the relative amplitude of a progressively more narrow band of frequencies around the cutoff frequency. Resonance is also known as "Q, emphasis, regeneration," or "feedback," the latter term descriptive of feeding back a portion of output into input. As "Q" or resonance is set progressively higher, the bandwidth of frequencies at the output becomes progressively more narrow. In some filter designs, high resonance settings causes production of (only) a sine wave at the cutoff frequency. Analog filters with this capability, given sufficiently high feedback (resonance), require no input signal to produce this sine wave, as resident noise in all electronic circuitry functions as the "input" waveform intensified by feedback. In this case, the filter behaves like a sine wave oscillator, and filter cutoff frequency is then the frequency control. • In a more general sense, in electrical circuitry, or in acoustic vibratory systems, resonance describes the tendency for a vibration (or electrical homologue) to achieve its greatest amplitude when excited, or driven at a particular "resonant" frequency, having a tendency to continue to vibrate for a time even when the driving force, or excitation is removed. • In the case of speech, vocal tract manipulations create resonant frequencies, producing a multiplicity of formants (emphasized frequency bands having specific center frequencies), that produce vowels. See formant.

response, input • See frequency response.

reverb • Hardware processor or virtual equivalent that adds reverberation to "dry," or non-reverberant audio signals, creating the effect of a selected virtual "room" in which processed signals apparently reside. See reverberation.

reverberation • Multiple echoes caused by reflections of sound waves, particularly within enclosed spaces such as music performance venues. Reverberation (also shortened to "reverb") time is the measurement of the decay (dying away) of these many echoes, i.e. reverberation in a room, from a maximum intensity level referenced as 0 dB, to an intensity level of –60 dB, i.e. a 60 dB loss from maximum. This loss (–60 dB) is a 1:1 000 000 (one to one million) power ratio. See (measurement unit) decibel.

reverse or reversed audio • Audio retrograde, sound reversal, reverse, or reversed audio, is analogous to retrograde (playing backward) of the notes of a musical theme. Reverse or reversed audio is achieved by playing a recorded segment of audio or digital sound file backward from its end (or some midpoint) toward its beginning. The chief perceptual characteristic of reverse audio is the reversal of the envelopes of individual sounds. See envelope. Retrograde is a music composition device that typically involves notation, whereas reverse audio is a sonic (sound) device facilitated by digital audio techniques.

RF • Radio frequency (RF) refers to a band of frequencies in the electromagnetic spectrum whose use is restricted to broadcasting radio waves. AM radio is spaced by 10 000 Hz (10 kHz) channels in the frequency band from 535–1605 kilohertz (kHz). FM radio is assigned to the frequency band from 88.1–107.9 megahertz (MHz). An RF carrier, or radio channel has a frequency far above human hearing, so the RF carrier signal's amplitude (AM radio), or frequency (FM radio) is modulated by the program signal of interest (music, speech, SFX). This program, or modulated signal is detected and recovered from the carrier by an AM or FM radio receiver (whichever is appropriate) and made audible. • Sound synthesis techniques such as high frequency amplitude modulation (AM) and linear frequency modulation (FM) have carrier (C) and modulator (M) signals–similar to broadcast radio, but typically both modulator and carrier are audio frequency signals in sound synthesis, i.e. both modulator and carrier might potentially be heard if monitored. In music synthesis using AM or FM, we typically monitor, i.e. listen to the carrier output–unlike broadcast radio, where any carrier frequency is far too high to hear. In broadcast radio, we hear the modulator signal, the result of demodulating the carrier signal–which has been made to "carry" the information of the modulator signal. See (modulation) AM. See FM.

ribbon controller • A signal generator that typically outputs two signals, in terms of MIDI: note on/off, and note number. However, "note number" produced by a ribbon typically offers finer resolution than the (eight bit, 0–127) MIDI keyboard note number featured in MIDI's first version (1.0). A ribbon controller uses two (2) bytes of data, facilitating a (perceptually apparent) continuous rather than discrete signal output. Therefore, control over parameters affected by the note number output of the ribbon controller behave in an apparently continuous, rather than discrete manner. • A ribbon controller is superficially known as an “alternate controller,” putatively intended to be used in lieu of, on in addition to a keyboard. However, the authors emphasize that any signal can be connected to any audio, control, and/or timing (ACT) input(s). See ACT. Avoid the stultifying (look it up, or we have people who will track you down and stultify you) limitation of thinking of a performance interface as a “controller,” at least prior to its connection to actually function that way. Consider how any so-called “controller” output might be otherwise connected, and use it creatively. (Experiment with it, “zipper neck!” See Young Frankenstein (1974), Directed by Mel Brooks.)

ring modulator • Balanced amplitude modulator (BAM), a processor that uses a four quadrant multiplier as its basis. The term “ring” dates from analog designs with a near-circular ring of four appropriately connected diodes. A diode is an electronic component that allows current to flow only in one (unipolar) direction. See ring modulation. See BAM. See (modulation) AM. See multiplier, four quadrant.

ring modulation • Form of amplitude modulation (AM), aka balanced amplitude modulation (BAM) using a four quadrant multiplier in which both carrier (C) and modulator input signals (M) are suppressed, i.e. neither input signal (C nor M) appears at the signal output. See ring modulator. Given input of sine waves at frequencies C and M, only the C+M and C–M sideband pair of output frequencies appears at the signal output. That is, only the sum and difference between input frequencies are present in the output. With more complex waves, each “C” partial interacts with each “M” partial, producing sum and difference partials for every possible pair, and a more complex waveform is produced. See (modulation) AM. See BAM. See multiplier, four quadrant. • As a ring modulator can output partials with frequencies that are not present at its inputs, it is classified as a “nonlinear” processor. See waveshaper, nonlinear. • Possibly because a RM (ring modulator) is a nonlinear device, and often embodied as a guitar player’s “stomp box” that features a local oscillator providing an “M” signal with a single tuneable, but static frequency, it is erroneously presumed by some to “ . . . produce [only] nonharmonics (inharmonics),” which would lead to “clangorous” (bell-like) sounds. But the output of a RM depends strictly on the signals at its inputs. If those signals are harmonic to each other, this device proves perfectly capable of outputting harmonics. Another example of the need to understand “how it works,” not merely “how to work it.”

RMS • Root mean square. See amplitude, RMS.

roll off • Cutoff filter slope that attenuates, or “rolls off” high/low frequencies above/below a particular cutoff frequency. The term is sometimes associated with the loss of lows “rolled off” by a high pass filter that has a gentle slope, or conversely and more commonly, highs “rolled off” by a low pass filter with a gentle slope. See frequency, cutoff. See filter slope.

room tone • The continuous background of unobtrusive sounds and noises in an ambient environment, particularly that of an enclosed space. Room tone is the sonic “picture” that a room “presents” due to the room’s response to heat, ventilation, and air conditioning (HVAC) noises, and other sounds in the room, in context of the reverberation characteristics of the room. See reverberation. A room's shape, dimensions, reflection versus sound absorption characteristics play roles in determining its peculiar room tone. • Room tone is an important element of producing sound for picture, and typically is altered creatively by the sound designer as characters on screen, or camera point of view (POV) move from one space or location to another. Credible room tone is an integral part of sound design for picture. See ambient. See resonance.

sample and hold • A processor circuit or module that can output a series of step signals that function like a changing series of constants. A sample and hold (S&H) can sense and capture, or sample the instantaneous level at its signal input, and hold and output that discrete level until the next sample is taken. A sample and hold (S&H) circuit can output a series of different discrete signal steps when samples of a dynamic continuously varying input signal are taken repeatedly. When the "clocking" signal is periodic, the principle is the same as pulse code modulation (PCM) audio signal sampling, whose design includes sample and hold circuitry. But, a freestanding sample and hold (S&H) module typically has a much lower bandwidth than PCM sampling. That is, a S&H module operates over a much slower span of sample, or clock rates than does PCM. The output signal of a periodically clocked S&H creates a metronomic, i.e. regularly timed series of steps whose values depend on the instantaneous level of the sampled waveform at the time each new sample is taken. • A sample and hold (S&H) module typically has an internal periodic clock, e.g. Low Frequency Oscillator (LFO) whose frequency can be set. Some designs provide an external clock input that responds as a threshold detector (much as the gate input on an envelope generator module). This timing input causes a new sample to be taken each time its threshold is exceeded, creating a series of held samples, or steps derived from the sampled signal connected to S&H signal input. Any step holds until another sample is taken. • An external "clocking" signal need not necessarily be periodic. For example, when a properly offset (biased) slowly fluctuating random signal (e.g. pink noise processed by a low pass filter set to a very low cutoff frequency) is connected to the S&H external "clock" input, the timing between steps at the signal output is aperiodic–decidedly not like a "clock." By definition, a clock has a periodic nature–it marks either absolute or musical time in regular increments of time, or pulses. See metronome. The S&H external "clock" input might be named more appropriately based on how it actually functions: it is a timing input–not necessarily a "clock" input. • The lesson is: so-called "clock" inputs in a modular sound synthesis system do not necessarily dictate input of a periodic signal, even though they may be used by many solely in this way due to convergent thinking. Learn about modular functions, input responses, and signal flow, and get creative by engaging in divergent thinking! • Sample and hold (S&H) circuitry is also used in less obvious ways. Analog voltage controlled, monophonic keyboard synthesizers, e.g. Minimoog, have S&H circuitry associated with each key. This analog S&H circuitry includes a capacitor, an electronic component that stores, i.e. holds a depressed key's unique DC (voltage) level. A monophonic analog keyboard holds the specified step value of the latest, lowest, or highest key when several keys are depressed, depending on the designer's choice of such "keyboard priority" options. In the typical case where an analog keyboard controls analog audio oscillator frequency, the corresponding pitch is maintained ("held") during the final envelope release (R) time of an envelope, when the operational (lowest or highest or latest) key played is no longer being depressed. Unfortunately, such analog keyboard S&H circuitry is a major source of instability, or pitch drift in vintage analog synthesizers, as the capacitor charge associated with the selected depressed & released key tends to slowly dissipate, or "leak" away over (short term) time, thereby causing the "remembered" analog level to "sag," or fall to a lower level. Typically, a pitch can go "flat" even while a key is depressed, but particularly when it is released and the note continues to sound during a lengthy release (R) time. See ADSR. Digital keyboards "hold" digitally encoded numbers in computer memory to represent specific key(s) depressed–note numbers in terms of MIDI. Digital keyboard designs do not suffer from "pitch drift" due to inclusion of "leak-prone" keyboard S&H analog components such as the capacitor. Cf. sequencer, step (a generator that outputs a series of programmable step signals).

sampling • Process whereby an analog (continuous) signal can be represented by discrete numerical magnitudes, stored in a computer, and reconstructed back to its original analog form. See sampling theorem. See (modulation) PCM.

sampling theorem • The mathematical basis and rationale that undergird sampling techniques, including digital audio. The sampling theorem states that any periodic analog signal can be sampled, digitized, stored in computer memory and then reconstructed into its original analog form without loss or distortion of data, provided that there are, at a minimum, at least two (2) samples taken per period of the highest partial (sine wave) of the sampled waveform. That is, the sampling frequency, or sample rate must be at least twice the frequency of the highest sine wave partial of a complex tone being sampled to avoid aliasing, or distortion of spectral information. See aliasing. • In honor of Harry Nyquist's contributions to this theorem, the nominally highest frequency that can be successfully sampled in any system (one half the sampling frequency) is known as the Nyquist frequency or simply, the Nyquist. The compact disc (CD), at its debut, featured a sample rate (sampling frequency) of 44.1 kHz, which yields a Nyquist frequency of 22.05 kHz.

schema • A systematic mental approach, or conceptual pattern in the mind that orders facts, ideas, and elements into a coherent structure that facilitates understanding of a field of endeavor. • According to the philosopher Emmanuel Kant, a schema is a means that allows application of mental understanding to account for or explain sensory experience. See Emmanuel Kant (he rarely takes visitors nowadays) for a better schema about the idea of schema, (and probably, a better definition of schema).

schematic • A diagram comprising standard electronic component icons (resistors, capacitors, amplifiers, etc.) that are configured into a usable electronic circuit.

Schmitt trigger • Circuit or module that senses the varying level of an AC signal at its signal input, and outputs a positive (+) DC step signal when that input signal exceeds the higher of two (2) predetermined or programmed input thresholds. When the input level is less than the second (lower) threshold value, the state of the Schmitt trigger is low and no signal (or zero signal) is output. When the input signal falls between the higher and lower thresholds, the present status (high or low) of the Schmitt trigger is retained. This differential between high and low thresholds provides added stability compared to a device with a single threshold value, e.g. in the case where a noisy signal might cause unwanted rapid switching back and forth between high/low states due solely to the noise component of the input signal. This step (DC) output signal is typically used to trigger, i.e. initiate the action of another module or circuit, and typically Schmitt trigger output might be connected to a timing input, e.g. of an envelope generator (EG). See threshold. See threshold detector.

SCO/SCF/SCA • "Signal controlled" (SC) modules, i.e. virtual equivalents of the primary modules of an analog voltage controlled synthesizer: voltage controlled oscillator (VCO); voltage controlled filter (VCF); and voltage controlled amplifier (VCA). "S" stands for "signal." See VCO/VCF/VCA.

scratchpad • Colloquial, or informal term for a buffer that comprises sectors of computer memory addresses reserved or dedicated to hold preliminary results such as most-recent computations, ephemeral (short-lived intermediate) results, or temporarily needed data. See buffer.

selector • Switch that has more than two positions. See switch, etc.

sequencer, step • A generator, the step sequencer signal output can provide a series or (potentially) a repeatable pattern of discrete signal levels, aka steps. Some designs provide an internal oscillator, or clock signal that dictates the period, thus the speed at which the step sequencer progresses serially through the finite number of available steps, aka programmable stations. Implementations of the step sequencer that provide an external clock input allow a variety of signals, including aperiodic signals to propel advances through these programmable stations. That is, the selected "clocking," or more accurately, timing signal need not necessarily be periodic, or "clock-like." • Each step sequencer "station" provides a single programmable direct current (DC) voltage step (analog) or constant (digital), comprising a series of programmable step signals as one progresses through the numbered stations. An attenuator associated with each station is used to program that station's step signal to the desired (±) DC level in analog terms, or constant in digital terms. In DSP (Digital Signal Processing) terms, a step sequencer outputs a series of numerical constants that typically have been calibrated within the system to produce a particular musical interval (e.g. half step) per each whole number constant, e.g. when controlling signal controlled oscillator (SCO) frequency. Often, adjacent whole numbers represent semitones. • The external clock input on a step sequencer is a threshold detector that advances the internal ring counter (analog sequencer circuitry) or algorithm (digital program) one station per each upward crossing of a default threshold value typically determined by the designer. In this case, when a signal connected to the sequencer's external clock input moves past 0 to any positive level (e.g. +1), the ring counter circuitry or digital algorithm advances once to the next station. In order to facilitate another advance, the input clock signal must first cross the same threshold downward, typically toward zero (0) in a digital system. The threshold detector is then "re-armed," and can sense the next upward transition, causing another advance. A single clock cycle in this scenario therefore involves two transitions through a threshold value, first upward–then downward. The external clock input is a threshold detector, specifically a timing input, but this "clock" input does not necessitate input of a periodic, or "clock-like" signal. For instance, an appropriately biased slow-moving random signal (e.g. low pass filtered pink noise) connected to the clock input, would cause the step sequencer to move through stations aperiodically, i.e. at random time intervals in the example given. The sequence of "steps" would be anything but metronomic. • Sequences with repetitive forward, backward, or bidirectional looping through stations may be executed using some designs. Some implementations may provide programmable non-serial or even random progressions through the numbered stations. Therefore, the "next" station whose programmed level is actually output is not necessarily the next higher numbered station, but may be subject to user definition or chance operations. Cf. sample and hold (a processor that outputs a series of step signals).

SFX • Acronym for sound effects (SFX), aka "sound FX." Sound effects are one of the three (3) major elements in the soundtrack of a film, the other two being dialogue (or narration), and music. Cf. foley.

shifter, frequency • Processor that raises or lowers the frequencies of all partials in the input signal linearly, by algebraically adding a selected frequency increment, not due to multiplication of those frequencies by a constant percentage, or factor. A frequency shifter changes the frequency ratios of the partials it processes, thereby altering the spectrum of the processed signal. This is usually heard as a change of timbre, or tone color. Cf. shifter, pitch. • For example, two partials (sine waves) with a frequency ratio of 3:2 create a perfect fifth (P5) musical interval. The frequency of the upper partial is 3/2, or a factor of 1.5 times the lower partial, (aka a ratio of 1.5:1). Given a lower partial of 440 Hz, the upper partial of a P5 is 660 Hz. When frequency is shifted up, e.g. 400 Hz, this selected frequency increment is added to each input partial. The resulting output then has partials at 1060 Hz (660 + 400), and 840 Hz (440 + 400), a new frequency ratio of 1060:840 (aka a ratio of 1.2619:1). The two partials of the input waveform, an unprocessed perfect fifth (P5), originally have a whole number, or integral 3:2 ratio, which is harmonic. The partials of this P5 harmonic ratio, when frequency shifted as in this example, now have a nonintegral (1.2619:1) ratio (based on frequency ratio of 1060:840), which is a new musical interval whose partials are inharmonic to each other. When a complex, (e.g. square) waveform with many harmonics is frequency shifted, the resulting output spectrum will likely have many inharmonics. Partials within this square wave are changed from harmonic to inharmonic, creating a clangorous sound, due to frequency shifting. See clangorous. Intervallic ratios among input partials are altered by frequency shifting. (The frequency shifted signal may also be perceived as having a different pitch as a result of frequency shifting, but this is a related byproduct of frequency shifting rather than the intended effect.) • The frequency shifter excels at changing pitched sounds with periodic spectra into clangorous, or bell-like sounds whose spectra contain various inharmonics. See clangorous. See (partial) harmonic. See (partial) inharmonic. Cf. shifter, pitch.

shifter, pitch • Processor that raises and/or lowers the frequencies of partials in an input signal logarithmically by multiplying those frequencies by a constant factor, rather than linearly by adding a selected frequency increment to each partial in an input signal. Cf. shifter, frequency. A pitch shifter raises or lowers the frequencies of input partials ratiometrically, i.e. by the same percentage. A pitch shifter therefore does not change the intervallic relationships, i.e. the frequency ratios of the partials it processes. The pitch of the processed signal is changed by pitch shifting, but processed signal spectrum is not changed (in an ideal design). Therefore, the pitch of the input signal is changed, but not its timbre, or tone color, at least when geometric waveforms are involved. (Pitch shifting can cause a change of "timbre" or "tone color" when formants present in the input signal are shifted, e.g. the human voice when pitch shifted on an ordinary sampling keyboard. See formant.) • For example, two partials tuned to a perfect fifth (P5) have a frequency ratio of 3:2 (three to two). The upper note of a P5 is three halves (3/2), or a factor of 1.5 times the lower note (aka as a ratio of 1.5:1). Given a lower partial of 440 Hz, the partial that is a P5 higher is 660 Hz. Given sine waves for each frequency, when such an input signal's pitch is shifted by a given interval, e.g. one octave, the frequency of each partial is multiplied by the same factor (e.g. 2), the ratio (2:1) of the given interval of shift (one octave). Such a ratiometric, or constant percentage pitch shifting operation, maintains the original 3:2 frequency ratio of the P5 (perfect fifth). It does shift it up one octave, to a ratio of 6:4. When a complex (e.g. square) waveform with many harmonics is pitch shifted, the output retains all ratios among partials, and the output spectrum therefore includes the same integer frequency ratios, or harmonics that originally appeared at the input. The frequency ratios among partials are preserved, but they are "shifted" by a constant percentage, e.g. 100% (a factor of (2:1), an octave in this example). Intervallic ratios of partials are therefore not altered by pitch shifting, so the input signal is shifted only in pitch–not in spectrum. That is, the "tone color" of input signals that lack identifiable formants, e.g. geometric waveforms, is not changed. See formant. Cf. shifter, frequency. • Pitch shifting with sophisticated algorithms for treating formants is used, e.g. to correct intonation in vocal tracks with minimal change of vocal tone quality (reflected by both spectrum and formants). See formant.

SI • The International System of Units is the accepted, modern form of measurement units and nomenclature based on the metric (base ten) system. SI units have been accepted worldwide in business and the sciences. The abbreviation SI is derived from the French Le Système International d'Unités.

sideband • One of a pair of partials (sine waves) produced by modulating one of the signal characteristics (e.g. frequency (FM) or amplitude (AM)) of an audio frequency sine wave carrier (C) with an audio frequency sine wave modulator (M). • If complex waveforms are used as carrier and/or modulator, each partial in the carrier waveform acts as a carrier that is modulated by each partial in the modulator waveform. The number of sideband pairs increases significantly when complex waveforms are used as carrier (C) and/or modulator (M). See FM. See C:M ratio. See index of modulation. See AM.

signal • In general, any gesture, action, indicator, sound, electrical condition, physical configuration, or display that might feature alternative conditions or more than one state used to communicate information. • In electronics, information or power transmitted by an electrical or electromagnetic signal or wave. In telecommunications, e.g. radio, television, etc. the data or information is "carried" by a higher frequency carrier signal, one of whose signal characteristic(s), e.g. frequency, amplitude, etc. is changed, or modulated by an associated modulation signal that carries the data of interest, e.g. speech, music, SFX (sound effects). See RF. See (modulation) AM. See (modulation) FM.

signal, ACT • Audio, control, and/or timing (ACT) signal whose function is determined solely by the type of input(s) to which that signal is connected in a modular sound synthesis system, as per the authors ACT dictum. See ACT.

signal, audio • See signal, ACT. See ACT.

signal, analog • A continuous signal that is not encoded, represented by steps, or communicated using discrete signal levels or numerical representations of same. See analog signal.

signal, aperiodic • Non-repetitive signal, i.e. one that has no discernible single unit of time during which a recognizable cycle is completed and repeated thereafter during identical ensuing units of time. • Aperiodic signals include random signals, e.g. white noise and pink noise; and nonrandom signals, e.g. the aperiodic signal produced by an envelope generator. See noise, pink. See noise, white. See ADSR. See EG segment. Cf. signal, periodic.

signal, bipolar • A signal or DSP representation that includes both positive and negative signal levels or numbers. • A waveform or signal that crosses zero level in order to exhibit both plus and minus (±) signal polarities or numbers. Cf. signal, unipolar.

signal, carrier • A signal whose characteristic(s) are modulated (changed) in order to "carry," or transmit information. See AM, FM, PCM, and Aunty Em. • In modules such as a multiplier, the carrier signal can "pass through" the processor. (In contradistinction, a modulator signal as such does not appear at the signal output of a processor.)

signal, control • Any signal connected to a control input. See signal, ACT. See ACT.

signal, digital • A signal represented in discrete, encoded steps, e.g. PCM or sampling. See digital signal.

signal, logic • The output on a logic module is constrained to have only one of two possible values, 0 or 1 at any given moment. The input of a logic module ignores magnitude changes of the input signal other than those immediately flanking its threshold value, typically located between 0 and 1. The input(s) on a logic gate (module) are threshold detector(s). See threshold detector. See logic gate. • In the authors' schema, technically there is "no such thing" as a "logic" signal in terms of signal functions. Logic gates, e.g. OR, NOR, AND, XOR, etc. have both signal input(s) and output(s) and are consequently categorized as signal processors. The output of a logic gate is ultimately connected to a “final” destination, or input that is an ACT (audio, control, timing) input. Logic gates (and their output signals) are therefore audio, control, and/or timing (ACT) modules, not some other special category unto themselves. There is plenty of logic to this–when you think about it.

signal, periodic • Repeating waveform or phenomenon that occurs, appears, or replicates its cycle or existence during each ensuing occurrence of a regular unit of time. Printed magazines and journals are called “periodicals” because they are published repeatedly at a regular time interval (monthly, quarterly, etc.) • The time (t) for one iteration, or cycle of a periodic waveform is its period (T). Period (T) and frequency (f) have an inverse relationship, that is $f = 1 / T$ and $T = 1 / f$. The period (T) for a signal with a frequency (f) of 1000 Hz is: $T = 1/1000 = 0.001$ second = 1 millisecond (ms or msec). In musical terms, audible periodic audio signals within the frequency range of human hearing are perceived as having a “pitch.” See pitch. Cf. signal, aperiodic. See clock.

signal, step • Signal having discrete level(s) or limited resolution numerical value(s), rather than infinitely divisible numerical and/or continuously variable analog level(s). Step signals are available at constant (bias), keyboard note number, keyboard gate, sample and hold, clock, and step sequencer signal outputs. • Even though a sampled audio file has many such discrete levels, or steps, it is not considered a “step” signal, as its steps are not intended to be discernible to the ear.

signal, timing • Any signal connected to a timing input. See signal, ACT. See ACT. Cf. clock.

signal, unipolar • Signal or DSP virtual representation that includes either, but not both positive (+) and negative (–) signal levels or numbers. Such a signal is known respectively as unipolar positive (+ only) or unipolar negative (– only). A unipolar waveform or signal typically includes zero (0), but does not cross zero level to become bipolar (±). Cf. signal, bipolar.

signal characteristic • Any identifiable aspect of a signal, e.g. frequency, waveform, spectrum, voltage, current, amplitude, polarity, etc. that can be determined by measurement. Signal characteristics are objective data based not on human judgments, as are sonic attributes, but rather upon observations or measurements made using objective means such as measurement tools and/or laboratory (test) instruments, e.g. oscilloscope, spectrum analyzer, frequency counter. See signal. Cf. sonic attribute.

signal generator • Device or module, e.g. oscillator, function generator, envelope generator, noise generator, etc. with a signal output, but no input that functions as a signal input. That is, a signal generator may have control and/or timing input(s), but no signal input, defined as an input that can potentially route a signal through that module to a corresponding signal output. (Control and timing inputs do not route signal(s) to the signal output of a module.) • Signal generators may be categorized by the characteristics of their output signal(s) as either: (1) aperiodic generators, e.g. noise source, envelope generator (EG); (2) periodic generators, e.g. function generator, oscillator. Either type of generator might be configured to produce either or both (1) continuous signals, e.g. sine waveform, ADSR envelope signal; or (2) step signals, e.g. "analog" or "note" sequencer, sample & hold, constant, etc. The preceding categories are based on the inherent features of the signal(s) generated. In contradistinction, the function of any signal generator, the way it is actually used, is defined solely by the functions of the inputs to which that signal generator's signal output is connected. See ACT.

signal processor • Device with both signal input and signal output that can alter, or process the signal that (potentially can) pass through it, e.g. filter, amplifier, attenuator.

silence • Is golden, particularly at concerts where adults actually prefer to hear music rather than some inchoate idiot screaming throughout the performance, (unless that idiot is one of the performers). • In musical terms, a written or performed rest, where no sound is initiated, putatively results in silence. Even in this case, previously performed sounds may reverberate in the performance venue during such a "silence." • In acoustics, only a highly absorptive and sonically isolated anechoic (without echoes) chamber creates conditions that approach absolute silence. That is, at most locations there is a room tone or an ambient soundscape. See room tone. See ambient. • See especially the writings and musics of John Cage, or Brian Eno for some compelling viewpoints on silence.

sinusoidal • Having the characteristics of a sine waveform or sin function in trigonometry (pronounce sine or sin like "sign"). This curve is expressed by the general equation: $y = a \sin bx$ where "a" and "b" are constants, aka coefficients, that represent amplitude (a), and frequency (b) respectively. See coefficient. See waveform, sine. See waveform, etc. Cf. constant.

slew rate • A measurement derived from the interval of time it takes for a device such as an amplifier to respond to a large, sudden change of level at its signal input or elsewhere within its circuitry. Slew rate is typically a measure of the time it takes a device to respond to a signal that starts at zero (0) level, and then "instantaneously" moves to full scale (FS), i.e. the largest possible output or level allowed by the design. When a square wave is the test signal, an amplifier with a slow or low slew rate deforms that waveform's theoretically instantaneous vertical rising and falling level changes, causing a potentially discernible loss of high frequencies at the output. • A so-called "lag processor" is designed to intentionally provide an adjustable increase of slew rate. Such a device essentially constitutes a Low Pass Filter. In fact, "glide" on an analog synthesizer results from routing the keyboard signal through a low pass filter prior to connecting that signal to the audio oscillator frequency control inputs. Analog "glide" is a treatment of the keyboard signal, not a feature of the oscillators being controlled. See portamento. See glide.

S/N ratio • Signal to noise (S/N) ratio, a quantification of the power of the largest input signal that causes "little" or "no" distortion in a system (e.g. audio), compared to the power of background noise that exists continuously in that system when no input signal is present. A S/N ratio (aka SNR) is expressed in decibels (dB), a unit of measurement that inherently represents a power ratio expressed logarithmically. For example, a 50 dB signal to noise (S/N) ratio specification means that any signal recorded at the upper limit (full scale) of the dynamic range of such a system is 50 dB more powerful than the resident (inherent background) noise, or noise floor of that system. Such a S/N ratio indicates that the background noise in such a system would be quite audible, particularly during quieter passages of the recorded program signal. (We have sensitive ears.) In this example, the recorded signal's largest allowable level stands at a 100 000:1 (one hundred thousand to one) power ratio to system noise, which is 50 dB greater than that system's resident noise level, or noise floor. The absolute level of resident system noise might not be stated explicitly, as the dB scale can express a dimensionless ratio of powers, where neither zero (0 dB) nor a voltage level is necessarily explicitly specified. See (measurement unit) decibel. See noise floor. See amplitude, full scale.

software • An invented, or artificially constructed language or subset of computer code whose commands cause operations in a digital computer. • A collection of such commands comprises a program, aka an "application," that facilitates or achieves some task, e.g. word processing or music sequencing. • Commands and protocols that constitute an operating system for the hardware of a digital computer is often referred to as "system" software. Cf. hardware.

solid state • Electronic component constructed using semiconductor (e.g. transistor), rather than earlier vacuum or gas tube technology. Solid state electronic components are categorized loosely as silicon technology, although many other materials may be involved in their construction. See IC (integrated circuit).

solo • Button, switch, or control on each channel of a recording console that selectively facilitates, i.e. isolates hearing of that channel by itself. Cf. mute. • A musical passage during which a single performer is prominent.

sonic • Relating to sound, or that which is audible to the human ear. Relating to the production, propagation, or use of (longitudinal) sound waves. See sound. See ultrasonic. See infrasonic.

sonic attribute • Any identifiable aspect of sound, e.g. pitch, timbre, loudness, that can be discerned solely by listening. Sonic attributes are subjective data based on human judgments made without recourse to objective measurement tools or laboratory instruments (oscilloscope, spectrum analyzer, frequency counter). Cf. signal characteristic.

sound • Objective vibrations and resulting waves that propagate (travel) through a solid, liquid, or especially a gaseous material elastic medium such as air. • Subjective responses, i.e. sensations experienced by a sentient being when those objective waves or vibrations stimulate the tympanic membrane (eardrum) and associated anatomical auditory mechanisms. • Normal human perception of sound has a bandwidth from 20 Hz–20 000 Hz. Those frequencies that fall below 20 Hz are infrasonic; those above 20 000 Hz (20 kHz) are ultrasonic. Cf. signal.

soundscape • The ambient (surrounding) sonic environment, by analogy to the visual environment of a landscape or seascape. See ambient. • A musical composition that is typically not traditional, perhaps without standard tonality (key) or time signature, where found sounds or recorded noises may predominate.

spatial • Pertaining to three dimensional space having axes labeled x,y,z. The spatial dimension of sound and music is perceived by identifying the apparent location(s) of various sound sources. • Spatial possibilities of recorded music have evolved from monophonic speaker configuration(s) that use one or more speakers producing the same sounds, to stereophonic speakers with left (L) and right (R) sonic differentiation, to surround sound using many speakers, e.g. 5.1 and 7.1 and 9.1 configurations that further increase the potential for locating sound in space.

spectral • Relating to spectrum, the collection of partials that comprise a complex signal, especially an audio signal. See spectrum.

spectrum • A group of frequencies related in some way, e.g. frequencies of the radio spectrum. • The constituent elements of a sound expressed as its particular collection, i.e. specification of individual partials (sine and/or cosine waves). • The term is used casually to refer to a line spectrum, a graphic representation of a waveform in the frequency domain. See spectrum, line. See domain, frequency.

spectrum, line • Graphical description of partials (constituent sine and/or cosine waves) of a waveform depicted in the frequency domain where the horizontal (x) axis represents frequency (linear or log), and the vertical (y) axis represents amplitude, typically in volts, percentage of full scale (FS), or power in dB. The line spectrum is so-named due to its depiction of a complex waveform as a group of parallel vertical line(s), each of which individually represents a single partial's frequency as per its unique location on the x axis, and amplitude due to that line's relative height on the y axis. The amplitude of a given partial (sine wave) is depicted by its height relative to some full scale (FS) value, e.g. 0 dB, or possibly simply an arbitrary representation of full scale such as +1.0. The polarity or phase of an individual partial is sometimes depicted in a rigorously technical line spectrum, in addition to frequency and amplitude. "Negative" partials may be represented "below zero level" in the vertical (y) axis, rather than above zero level, i.e. in the opposite direction from that of "positive" partials. In this context, a negative partial is illustrated graphically as having the opposite polarity of a positive partial. Alternatively, negative partials may be marked simply with a negative (–) sign or red numbers. In some cases, negative partials may be depicted to the left of a "zero" frequency point in the middle of a bipolar (±) horizontal (x) axis line.

spectrum, potential • Spectrum (group of partials) that is produced when the extent (depth) of modulation of a carrier is at its greatest, aka full scale (FS). • The spectrum that a particular carrier to modulator (C:M) frequency ratio can produce when the index of modulation (depth) is at its maximum, during rapid amplitude modulation (AM) or rapid frequency modulation (FM). A zero (0) modulation index produces no sidebands, therefore none of the "potential" spectrum (all possible partials) that can be produced by a modulation index set to maximum (full scale) or portions thereof at some intervening value. However, the locations of the frequencies of partials in any potential spectrum remains a function of the selected C:M frequency ratio, whether any of that spectrum is produced due to modulation, or not. The spectrum is "potentially" available should modulation be effected. The C:M ratio dictates its makeup. See spectrum. See sidebands. See FM. See (modulation) AM. See FM modulation index.

spectrum analyzer • Test device hardware or computer software that analyzes an input signal in the time domain, i.e. its waveform, and outputs a representation (line spectrum) of that waveform's constituent partials in the frequency domain. In the frequency domain each partial's frequency, amplitude, and in some cases, phase or polarity are shown graphically using a vertical line whose height represents that partial's amplitude relative to full scale. See full scale. A spectrum analyzer typically displays a dynamic histogram (graph with data depicted as vertical bars or lines) of the spectrum that details the amplitude of each partial on the vertical (y) axis, while illustrating its frequency its frequency on the horizontal (x) axis. See spectrum, line.

speed • Speed is the rate of change of distance traveled in time in a straight line or a curve that is constant. The rate of change of the displacement of a body that moves in time is called velocity. Velocity (v) is illustrated by a vector that indicates both magnitude and direction. The terms "speed" and "velocity" appear interchangeably in casual use, but the technical distinction remains valid.

speed of sound • Sound waves travel at a speed depending on several factors, particularly the density and elasticity of the material medium through which they propagate, or travel. • The speed of sound in warm, dry air is 344 meters per second, i.e. 344 m/s @ 20 ° Celsius. The change (delta) in the speed of sound in air due to changes of ambient air temperature is 6 meters per second per each change of 10 degrees Celsius (6 m/s per 10 ° C). Sound travels faster in warmer air, and slower in cooler air, when all other factors are held constant. See ambient. • The speed of sound through a medium other than air varies widely, based primarily on that medium's density and elasticity. For instance, in water the speed of sound is 1410 m/s. For steel it's 5100 m/s.

SPL • In audio engineering, sound pressure level (SPL) is a measurement of short-term (audio frequency) deviations relative to ambient air pressure. Sound waves create such SPL deviations. (Barometric pressure, a factor used in forecasting weather, typifies longer-term deviations of ambient air pressure. The local barometric pressure in the surrounding area is aka ambient barometric pressure.) See ambient. • There is a square relationship between sound power, i.e. intensity (I) and SPL, expressed as: $I = p^2 / \rho v$ where "I" is intensity, "p" is sound pressure, "ρ" (the Greek letter "rho") is the density of air, and "v" is the speed of sound. Some authorities prefer " L_p " in lieu of "SPL," and use the attendant formula for pressure ratios: $L_p = 20 \log p/2\text{x}10^{-5}$ where "p" is pressure, and " $2\text{x}10^{-5}$ " is the standard reference pressure in Newtons per square meter, equivalent to 20 micropascals (0.000 020 pascals). • SPL may be measured using a meter comprising a calibrated microphone (mic), amplifier, selectable A, B, or C filter network, and a display that indicates decibel (dB) levels. Such SPL readings range from "intolerably intense" (jet plane takeoff @ 60m ≈ 120 dB), to "minimal intensity," i.e. barely audible (rustling leaves ≈ 10 dB). See (measurement unit) decibel.

static • Electrical interference that causes noise in a broadcast medium or communications system, typified by intermittent or random bursts of crackling or sputtering sounds. • Motionless or fixed, the opposite of dynamic, in regard to a signal characteristic or parameter value. The static setting of a constant (bias or offset) may be changed by the user, but this level or value does not subsequently change of its own accord. In contradistinction, a dynamic signal from, e.g. an oscillator, noise source, or envelope generator, has amplitude values (levels) that change of their own accord, requiring no further intervention from the user, once initial values are programmed (e.g. bias of oscillator frequency) or initiated (e.g. envelope generator A, D, and R, segments, once "triggered," or gated). Cf. dynamic.

steady state • Sound element that constitutes the sustained, or "middle" part of a sound, following the attack transients (beginning) and preceding the decay, or release transients (ending). The steady state typically exhibits frequency and amplitude changes of constituent partials that are typically less dynamic than the transients (attack & decay) of the same sound, hence "steady" state. Cf. transients, attack. Cf. transients, decay. See dynamic.

stereophonic • An audio recording & playback system that uses, at minimum, two transducers (loudspeakers or headphones) capable of reproducing different aspects (tracks or channels) of recorded sounds or a mediated musical performance. That is, different sound event(s) can appear independently in left (L) and right (R) speakers simultaneously. Or the same sound event(s) can appear at different levels in left (L) and right (R) speakers. Stereophonic (L-R) speakers are used to transduce different sounds or levels independently, e.g. as mixed down from tracks of a multitrack recording. A stereophonic system has two channels for transmission of audio signals from source to the transducers or speaker(s). See pan pot. Cf. monophonic.

stop band • See band, stop.

subjective • Existing in the mind, with no corroborating or supporting evidence provided by an objectively measured external reality. Not directly verifiable by measurement, but deriving from opinions, feelings, or some arbitrary plebiscite (vote) or consensus regarding some imagined external reality that presently can neither be verified nor disproved, e.g. whether extraterrestrial intelligence exists. Cf. objective.

subsonic • From "sub," for below, and "sonic," meaning sound, therefore traveling at a rate slower than the speed of sound. The term relates primarily to aerospace vehicles traveling slower than the speed of sound, which is approximately 768 miles per hour (mph). The term "subsonic" is sometimes used, but is inappropriate when describing frequencies below the lower frequency limit of human hearing (20 Hz). The appropriate term for acoustics and audio engineering is infrasonic. See infrasonic. Cf. ultrasonic.

subwoofer • A transducer, i.e. speaker or radiator in a high fidelity sound system designed to optimize production of the very lowest frequencies (<100 Hz) in the audible band of frequencies.

summation, algebraic • See algebraic addition.

summing node • Electronic circuitry or software algorithm that algebraically (±) sums two or more signals, and makes the resulting signal or mathematical summation available, typically for internal use within a sound synthesis module. The term summing node is particularly associated with control inputs, both internal and external, on signal controlled (SC) modules (SCO, SCF, SCA, etc). Multiple signals connected to such inputs that control a particular parameter, e.g. oscillator frequency, filter cutoff frequency, amplifier gain, etc. are added algebraically at the summing node within the module to which these control signals have been connected. Applied control signals for a selected parameter typically sum; they linearly and algebraically (±) add in most modular sound synthesis systems. See algebraic addition. • Hardware modular sound synthesis systems that use banana plug patch cords that can be "stacked" by placing the plug of one patch cord into the jack on the back of another patch cord do not conform to this dictum of simple linear algebraic (±) addition of control signals. Use of several banana patch cords to route several signals to a single (control) input without use of a summing node might be construed as an attempt at "mixing on a wire." "Adding" several signals by placing those signals on a single wire (e.g. several interconnected patch cords) yields nonlinear, possibly unpredictable results. A similar nonlinear summation of signals is also effected by routing more than one signal into a multiple–a module that comprises three or more interconnected jacks designed to provide replicas of a single signal connected to one jack. Neither a wire nor a multiple module functions effectively as a "mixer," in the ordinary sense of that word. See mixer. Cf. multiple. • Both linear signal summing, aka "mixing," as well as nonlinear signal summing as described, are often referred to using the eighteenth letter of the Greek alphabet sigma, transliterated as "s." In math, the capital letter sigma (Σ) means summation. The lower case sigma (σ) represents the parametric standard deviation, a measure of the dispersion (deviation from some average or central value) of an entire population (rather than a sample, or limited subset of that population), a concept in statistics.

supersonic • From "super," or above, and "sonic," meaning sound, therefore meaning travel at a rate faster than the speed of sound. The term relates primarily to aerospace vehicles traveling faster than the speed of sound, which is approximately 768 miles per hour (mph). The term "supersonic" is sometimes used, but is inappropriate when describing frequencies above the higher frequency limit of human hearing (20 kHz). The appropriate term for acoustics and audio engineering is ultrasonic. See ultrasonic. Cf. infrasonic.

surround sound • Any of a number of speaker playback configurations (and corresponding recording–mix configurations) using multiple speakers to allow greater latitude in positioning sound(s) in three dimensional space. Configurations typically have added "rear" speakers located behind the nominal stereophonic "front" speakers listening position, and subwoofer(s) to handle the lowest frequencies played. Such surround sound is referred to as "5.1" or "7.1" to indicate the number of full bandwidth speakers ("five" or "seven"), and sub-bass ("point one") capability. Electronically produced versions of surround sound, e.g. "2.0" are effected using DSP, and require only a standard pair of (stereo) speakers. See stereophonic.

sustain • That portion of a tone that may be performed or held indefinitely, i.e. the "steady state" rather than the transients of a sound. Cf. transients, attack. Cf. transients, decay. • The third segment (S) of an ADSR envelope generator output whose timing typically depends on the presence of a "gate on" condition at the gate input of the envelope generator (EG). In this context, a sound may be "sustained" indefinitely. See ADSR. • The term "sustain" is also used to describe use of the "sustain" pedal on a piano, which allows string(s) to continue sounding following release of the actual key(s) depressed to play those notes. In the case of the piano, sustain does not describe a situation where a tone will sound indefinitely.

switch • Bi-state mechanical, electronic, or virtual device designed to enable or disable, start or stop, open or close, make or break, or select between two alternatives, such as the connection between two electronic hardware circuits or two virtual devices. • Some switches have more than two states, but it is preferable to call such switches "selectors" to avoid confusing them with bi-state switches. See selector.

switch, momentary • Device whose action functions or result occurs, only during the instantaneous time, or "moment" during which the switch is physically held in one of its two positions (i.e. typically, on).

switch, push-push • Device that changes alternately from one status to the other each time that switch is activated (pushed). • Some push-push switches are designed to "scroll" through a given number of selectable alternatives. The authors suggest that the term "selector" be used to indicate such many-alternative devices, rather than the term "switch," which would be reserved to describe a two-state device.

switch, software • Virtual equivalent of a hardware switch. See switch.

symbol • Representation of an idea, variable, known value or constant, or some specific thing in a particular context (e.g. mathematics) using a printed, iconic, or written sign or character. See symbols, magnitude. • An abstraction that represents a physical object.

symbols, alphanumeric • The set of letters, symbols and numbers in a language, aka "alphameric." The essential distinction between numbers that are alphanumeric data, from the numbers in arithmetic, is that calculations are not intended for alphanumeric symbols. Numbers in the context of alphanumeric symbols function essentially as naming devices, such as the subscripts in the decibel (dB) formula: $10 \log (W_1 / W_0)$, where each of the subscripts (lowered numbers) " $_1$ " and " $_0$ " identifies a different quantity in watts (W) generically. Such subscripts typically do not represent a quantity that can be manipulated mathematically. The terms "W_1" and "W_0" are spoken as "W sub one," and "W sub zero" respectively. The use of such subscripts allows handy and unequivocal (clear-cut) identification, or naming of individual entities in a group of similar items.

symbols, logic • The set of symbols used in Boolean algebra, symbolic logic, to exercise propositions and make logical operations. Any competent text about symbolic logic explains these symbols.

symbols, magnitude • Indicators of comparative quantity or numerical size, e.g. less than (<); greater than (>); and equal to (=), or equality. There is also (≈), which means approximately equal to. For example, 3 < 9 (three is less than nine), and conversely 9 > 3 (nine is greater than three), and 3 = 3, and 9 = 18/2. The equality sign (=) can be used to express a known value, e.g. v = 344 m / s @ 20° C (the velocity (v) of sound in air at "room" temperature). • A memory aid: when interpreting (<) or (>) symbols, move from left to right as in reading words in English, and say "less than" when the small, closed end (<) appears first, or "greater than" when the large, open end (>) of the symbol appears first. • Size (magnitude) does matter. See (but do not bend, spindle, mutilate, or ever mess with) Godzilla.

synchronization • Alignment of several events or streams of data in time (t), or pertaining to events progressing at the same rate or occurring simultaneously. Relative "sync" or "synch" (synchronization), e.g. of sonic events to a musical tempo depends on establishing a timing reference, or "beat" using a metronome, ensemble conductor, etc. that is likely not absolute, and very likely not precisely periodic. See periodic. See tempo. See metronome. Absolute synchronization is based on accepted or standardized subdivisions of the passage of "real" time, such as hours, minute, seconds, etc. The timing reference (clock) in this case is typically rigidly periodic. See address, SMPTE time code.

synthesis • The composition, construction, or constitution of a whole due to the independent manipulation or control of its basic, atomistic, irreducible, or constituent elements. Sound and music may be synthesized electronically, as per this definition, using synthesis engines or techniques. See synthesis, etc. below. The term "synthesis" is particularly apropos to additive synthesis, linear frequency modulation (FM), and subtractive synthesis engines. See synthesis, additive. See synthesis, subtractive. See synthesis, FM. • The term "synthesis" rightly involves actions, and/or programming techniques that are atomistic–rather than holistic. This implies having a choice over which elements (modules) in a system that may be programmed to create a given sound, e.g. which type of filter might be utilized in a subtractive synthesis system. In most true synthesis techniques, programming of the various parameter(s) on selected modules is done independently of other parameters in the system, a hallmark of how "synthesis" is defined, at least by the authors. Sound generating techniques such as physical modeling may indeed involve programming of individual parameters, but a given parameter's value typically cannot be maintained independently, i.e. without causing changes in the value(s) of associated parameters–changes that may be difficult to anticipate. That is, physical modeling is based on mathematical descriptions of a physical entity, which necessarily dictates a holistic paradigm, in which case many parameter values in the system are interdependently dictated rather than chosen. Most of these parameters interact with each other in complicated ways that are hard to predict accurately without recourse to rigorous mathematical calculation. Although the math of physical modeling is deterministic, actual parameter interactions are not readily apparent to those who program them, even after significant investment of time to learn such a system. These interactions are therefore essentially beyond the immediate control of the programmer, even though a variety of parameters may be programmed, seemingly independently. If "synthesis" is defined as providing even a modicum of independent control over constituent elements (as most "synthesizers" afford, in the authors' view), then some techniques such as physical modeling do not qualify as "synthesis," and instruments based on these techniques should not be called "synthesizers." See synthesizer. See (synthesis) physical modeling. • Nor should holistic performance instruments, e.g. theremin, Trautonium, ondes Martenot be considered "synthesizers," as their use does not involve significant "programming" of atomistic sound elements. These instruments are used in performance, in the long tradition of acoustic musical instruments of the orchestra. • The term "synthetic" clearly denotes artificial construction, meaning "not natural," whereas "synthesis" rightly should not have such a connotation–to the extent that any musical instrument might be considered "natural." Guitars may be made of wood, but they do not grow on trees. A guitar may be constructed using organic materials such as wood, but the finished product per se is by no means "natural" due to this basis for its construction. All musical instruments, including the guitar, are highly contrived artifices designed to provide pleasure for human beings. An "acoustic" guitar is no more "real" or "natural" than a digital sampler, electronic music synthesizer, or a silicon-based digital computer. In fact, the chemical element silicon (Si) is one of the most plentiful substances on Earth. It occurs naturally in clay, granite, sand, and some minerals. This, however, does not make computers or other silicon-based devices "natural," any more than making guitars out of wood makes them "natural." In some sense, all musical instruments are "acoustic," to the extent that they are used to make sounds, rather than to function as stands for potted plants. Some instruments vibrate loudspeakers; others vibrate other kinds of materials. We tend to call the latter kind "acoustic." In the sense that an instrument causes vibrations, then all musical instruments might be considered "acoustic" instruments. In the sense that some instruments require use of electricity as an integral part of their design, then there are indeed "electronic" or "electromechanical" musical instruments.

synthesis, additive • Synthesis system, or "engine" that facilitates control over several signal characteristics of individual partials, particularly the individual amplitudes of a group of sine waves that are subsequently summed to create a single complex wave. Each sine (or cosine) wave that is a constituent element, or part of a complex wave is known descriptively as a partial. • Rudimentary additive synthesis would provide alterable, but static, i.e. non-dynamic amplitude control over individual partials. The frequencies of such partials might also be restricted to harmonic relationships, i.e. frequencies related by whole number ratios, e.g. as those found in a harmonic series. For an example, inspect the drawbars (faders) that govern amplitudes of harmonics generated by the tone wheels of the early (1935) electric organ designed by Laurens Hammond. •

Virtual additive synthesis designs may provide independent, dynamic control of each partial's frequency, amplitude, and (in some cases) phase, using associated envelopes programmed by the user. In this context, partials may be caused to dynamically vary in amplitude, and may also be varied between harmonic and inharmonic relationships. See synthesis, etc. See fader. See envelope. • Any synthesis technique that creates a more complex waveform by summing or transforming simpler waveforms, e.g. Walsh functions, might be considered additive. See Walsh functions. • Cf. synthesis, subtractive. Subtractive synthesis is a technique or synthesis engine that uses filter(s) to attenuate (reduce) or amplify (boost) partials in selected frequency band(s) of a complex waveform. See frequency band.
synthesis, AM • Technique or sound synthesis engine that relies heavily on rapid amplitude modulation (AM). AM is a rapid (audible frequency) modulation technique, whereby the amplitude of a carrier signal is modulated by the frequency and amplitude of a modulator signal. When both signals are sine waveforms, a single pair of sideband frequencies can be produced. See (modulation) AM. See BAM. Cf. FM. In AM synthesis, both carrier and modulator frequencies typically lie within the audible bandwidth (20Hz – 20kHz) • Amplitude modulation (AM) is the telecommunications technique first used popularly for radio transmission, in which case the carrier signal is invariably a fixed frequency in the ultrasonic (higher than audible) range, and modulator signals comprise the audio (music and speech), aka program signal that is transmitted. In the case of sound synthesis, and unlike radio, typically both carrier and program signals lie within the frequencies of the audible window (20Hz–20kHz).

synthesis, FM • Technique or synthesis engine that relies primarily on (linear) frequency modulation (FM). FM is a rapid modulation technique whereby the frequency of a carrier signal is modulated by a modulator, or "control" signal. Even when both signals are sine waveforms, many pairs of sideband frequencies may be produced depending on the amplitude of the modulator signal. The number of sideband pairs produced when sine waves are used depends on the index (depth) of modulation. See FM. See FM carrier. See FM modulation index. See FM modulator. See FM operator. See FM potential spectrum. Cf. AM. • FM is a telecommunications technique developed following AM, both of which continue to be used for radio transmission. The FM carrier signal is invariably a fixed frequency in the ultrasonic (higher than audible) range, and modulator signals comprise the audio (music and speech), or program signal that is transmitted. In the case of sound synthesis, typically both carrier and program signals lie within the frequency span of the audible window (20Hz–20kHz).

synthesis, granular

synthesis, Mr. Potato Head • Term coined by our colleague Mr. Chris Noyes, to describe a particular synthesis engine that concatenates (appends) a sustainable geometric (e.g. pulse) waveform onto a short digitally sampled "attack" segment. The initial digital audio sample file provides verisimilitude ("realness," or believability) due to its organic recorded attack transients. The geometric waveform that follows provides a means of reducing the bandwidth (data required) to make a credible sound that can also be sustained indefinitely, without recourse to "looping" the tiny initial sample file. This engine is essentially a hybrid sample playback-synthesizer. The need for such a technique is essentially of historical interest due to current availability of massive amounts of computer memory (to support longer samples), and much faster computer throughput generally. However, the success and economy of this technique does point out the importance of transients in creating believable, nuance-filled sound synthesis. See transients, attack. See transients, decay. See concatenate.

(synthesis) nonlinear waveshaping • Sound processing system that provides means for routing one or more waveforms into a processor designed to “distort” or deform, i.e. alter waveform(s). The term “nonlinear” indicates that this type of processing is specifically designed to output partial(s) not found in the original (input) waveform(s). Such waveshaping is often provided as a “post” processor, at the output section of a synthesis system such as frequency modulation (FM), subtractive, or additive. Nonlinear waveshaping is a holistic, rather than atomistic technique, and as such it is dubious to refer to such processing as “synthesis,” defined as the composition of a whole due to independent manipulation of its individual, constituent, i.e. atomistic elements. See waveshaper, nonlinear. See (modulation) AM.

(synthesis) physical modeling • A necessarily virtual means of sound production where sound parameters are interrelated so inextricably that they cannot be manipulated independently from each other without causing complicated and essentially unforeseeable interactions. This is endemic to the nature of any actual physical sound generator, or any musical instrument that is mathematically modeled on such physical systems, where a change of one parameter, e.g. wind instrument tube length, necessarily interacts with many other parameters, e.g. scale intonation, timbre, etc. in complicated ways that may be difficult for the ordinary user to predict. In a word, the physical modeling paradigm reflects the complexity of the physical world. While it is true that a physical modeling system may provide a variety of parameters that may even superficially “look” like a synthesis system, physical modeling should not be considered “synthesis” in the classic sense, where hardware or software provide individual and independent control over various sound parameters, e.g. frequency, attack time, filter cutoff frequency, etc. without causing unforeseeable interactions.

synthesis, subtractive • Synthesis system that features use of various filters to “subtract,” or attenuate bands of partials in complex waveforms, e.g. pulse, sawtooth, digitized sound files, etc. • Although early (voltage controlled) synthesizers feature subtractive synthesis, and happen to have been implemented using analog circuitry, the terms “analog” and “subtractive” should not necessarily be intertwined. That is, analog involves options concerning electronic circuit design, and subtractive refers to a particular sound synthesis technique or system that features prominent use of filters. Therefore, “analog synthesizer,” if meant to indicate a synthesizer with a subtractive synthesis engine, is a misnomer. But “analog synthesizer” is an appropriate appellation when meant to indicate a synthesizer with analog (continuous rather than discrete signals) sound generating and processing circuitry. An analog synthesizer might be designed to provide any of a variety of synthesis engines or techniques such as subtractive, additive, FM, AM, etc., any of which are implemented using analog circuitry that deals with analog signal generators and processors. See analog. Cf. digital.

synthesis, table-lookup

synthesis, vector

synthesis, Walsh function

synthesizer • A modular system or set of parameters grouped into modules that provides means of constructing, or constituting sound as a whole, due to the process of independently manipulating or controlling sound's basic, atomistic, irreducible, or constituent elements. These elements of sound are often parsed into groups of parameters. See parameter. • The use of the term "synthesizer" is dubious when referring to holistic devices such as early electronic musical instruments (e.g. ondes Martenot, theremin, trautonium), simple read only memory (ROM) sample playback devices, and other instruments (e.g. Fender Rhodes and Wurlitzer electromechanical pianos) that afford little or no independent programmable control over the constituent elements of sound. For example, Russian physicist Lev Terman's Ethervox, or theremin (1919) is not the "first synthesizer," nor is it even a precursor to the synthesizer. Such instruments aren't synthesizers at all, they are electronic musical instruments that one plays in the virtuosic instrumental performance tradition a la the symphony orchestra. In particular, the theremin affords very little programming control over the constituent elements of sound–it's a holistic performance instrument. Only those devices that generate sound due to synthesis of sonic elements could reasonably be called "synthesizers." See synthesis. • Choice of language is important if we are to communicate effectively. "Lightning bug" and "lightning" share similarities, but one will charm you, and the other will harm you. (Warning: do not ground yourself when lightning bugs are buzzing around in the immediate area.) • A synthesizer may be modular, comprising many separate, identifiable individual generators and processors that are "patched," i.e. physically or virtually connected together to create a sound-making configuration. Or, such modular functions may be integrated into a "hard-wired" package, where patching possibilities will likely be severely limited, but programming of individual parameters nevertheless remains possible (e.g. the Moog Minimoog, a hard-wired, programmable, subtractive synthesis engine, analog hardware synthesizer). A synthesizer may be a hardware device, or it may be virtual–in the latter case the generators and processors exist solely through the agency of mathematical computations within a digital computer.

tape echo • Repetition of recorded sound(s) caused by mixing playback (repro head) output back into the active recording input (record head) of an analog tape recorder, particularly on the same track. This creates tape echo, aka "slapback," due to the feedback loop from playback to record head. • This feedback loop creates a number of iterations (echoes) of each sound initially recorded, based on the gain provided in the feedback loop. Negative feedback gain (less than unity) facilitates a series of echoes with progressive decreases in signal amplitude until a given sound event eventually disappears. Positive feedback gain (greater than unity) facilitates a series of echoes that grow more intense with each iteration, eventually increasing to a signal intensity that saturates the tape, causing severe distortion. • The time (t) between successive echo(es) of any recorded event is determined by tape speed and the distance between record and repro (playback) heads. • "Tape echo" effects can be produced easily using a digital delay, in which case parameters (e.g. delay time) can be manipulated arbitrarily and independently. Feedback patching between tracks, and gain considerations are functionally the same as with analog audio magnetic tape echo.

temperament • The set of frequency ratios used to adjust the various frequencies that comprise a musical "scale" of discrete pitches used primarily to make tonal music. Twelve tone "equal" temperament is an equitempered scale with 12 half steps of the same size, all based on the ratio of approximately 1.05946:1 (the twelfth root of 2, or $2^{1/12}$). On the other hand, "just" tunings or temperament are predicated on intervals derived from integer (whole number) ratios, e.g. 3:2 for a "perfect fifth;" 4:5 for a "Major third," etc. A set of such "just" intervals can also comprise a harmonic series. See harmonic series. There are many temperament systems in music from various historical epochs. See ratio. See cent.

temperature •

tempo • Speed at which a musical passage with a succession of regular pulses progresses. • Musical subdivision of absolute (real) time (t) into a number of "pulses" or "beats" per minute. A mechanical or electronic device called a metronome can present these subdivisions audibly and/or visibly, providing various speeds (tempi, plural of tempo) in metronomic markings (MM), i.e. pulses per minute. For example, "march" tempo is 120 MM. Such metronome pulses can be associated with any note value (quarter, eighth, etc.) at the composer's discretion. See metronome.

temporal • Relating to objectively measured time (t), i.e. the passing of real time as reported by chronometers and clocks. • Relating to the subjective experience of time, as in perceiving musical time or tempo. Cf. tempo.

terminal • One of three classes (generator, processor, terminal) of devices in a sound synthesis system, a terminal is a beginning point or end point, i.e. one extremity or the other in terms of signal flow into or out of a system. In sound synthesis systems, a terminal is an input-output (I/O) access or port, e.g. external audio port (input), audio bus (output). A terminal is not a module per se, but is a type of device in a modular system. Cf. (module) generator. Cf. (module) processor. • A terminal is a point for making a connection in an electrical circuit (e.g. automobile battery positive and negative terminals) or a virtual representation of circuits or modules.

threshold • Generally speaking, a point, signal level, or numerical value that, when exceeded, causes or initiates an event, signal state, or other condition or experience. A boundary level that separates or differentiates between different conditions, responses, or operations. • In sound synthesis modules, a threshold is a level expressed as a number, power (dB), or signal amplitude characteristic such as voltage, used to determine whether an associated input signal's level exceeds or does not exceed that particular threshold, or boundary level. See threshold detector.

threshold detector • An input that senses level in a binary (two-state) way. Internal circuitry or algorithm in a module or system that conditionally produces one or the other levels of a bi-state signal. The status of this bi-state "high or low" signal is determined by whether the instantaneous level of a connected input signal exceeds (high), or does not exceed (low) the programmed level or number that represents the threshold. In many designs, any positive (+) value, e.g. +1 exceeds a threshold set conceptually between zero (0) and one (1), thereby producing the "high," or "logical yes" signal used to produce the now of a particular now versus not-now binary conditional scheme. In this case, zero (0) or any negative (–) number would produce the "low," or "logical no" signal used to determine the not-now of this binary conditional scheme. The signal produced by a threshold detector may be used within the module to alter how that module processes an associated input signal (e.g. noise gate input threshold); or it may cause a module to start production of an output signal (e.g. envelope generator gate input). See threshold. See noise gate. See generator, envelope. • For example, the typical envelope generator (EG) module has a timing input that comprises a threshold detector, variously called gate input, trigger input, or simply gate in. For the envelope generator typical of most virtual systems, any positive (+) signal connected to the EG gate input causes that envelope generator to start producing the first segment of its signal, called Attack. When the magnitude of this connected timing signal subsequently becomes zero (0) or negative (–) following the initial high (+) condition, the EG starts producing the final segment of its signal, called Release, in response to this low (zero (0) or negative (–) timing input signal) condition. In this context, the high versus low terminology for the timing signal connected to the gate input seems more appropriate that "on" versus "off," and possibly is more revealing even than "logical true" versus "logical false" terminology. That is, the envelope generator responds to either of the two gate input (threshold detector) conditions by starting an envelope segment (Attack or Release). Both high and low input signals result in an "on" condition–each of which initiates an action, albeit the second action always follows the first. We might say that the initial high state at the EG Gate Input enables the envelope generator to respond to any ensuing low state at that input. There may be any number of intervening EG segments produced between the first (Attack) and last (Release) segments that are started respectively due to high then low signals present at the gate input. Typically, these "additional" EG segments occur during the "high" state at the Gate Input. See ADSR.

timbre • Tone color, or timbre (pronounced tam´bur, or tom´bur) is a holistic, highly subjective evaluation of objective audible audio signal elements that include spectrum or waveform, envelope (transients), vibrato, tremolo, etc., as well as a subtle evaluation of the musical performance per se. • The objective frequency domain depiction of a steady state audio signal, known as its line spectrum, might be equated to the corresponding tone's timbre. A rigorously accurate line spectrum shows the polarity or each partial as well as its frequency and amplitude, constituting an excellent objective description of a signal deconstructed into its constituent elements, or partials. However, the ordering or timing of the appearance of individual partials that constitute this line spectrum, particularly attack and decay transients, can obscure the relationship between a static line spectrum (steady state sound) and an evolving timbre. For example, audio played in reverse creates a different perception of timbre than the same passage does when played normally, yet the line spectra for the steady state portion of those sonic events per se remains the same. See transients, attack. See transients, decay. • Timbre is often equated with waveform, the time domain depiction that differentiates two periodic signals whose frequency, amplitude, and phase are identical. However, waveform is possibly an even less reliable indicator of timbre than is a line spectrum. A change of either polarity or phase of a single partial in a complex tone typically causes a radical change of the complex tone's corresponding waveform. But, due to the ear's relative insensitivity to such polarity and/or phase relationships, any consequent change of timbre may be undetectable by ear. (There are cases, such as "klang" tones featuring inharmonics or low frequency tones, where such changes of phase or polarity of partials are detectable by ear). However, it remains true that radically different waveforms may share essentially the same perceived timbre. • It is well known that a sine wave of constant amplitude has a different loudness at different frequencies, as revealed by Robinson-Dadson (and earlier Fletcher-Munson) equal loudness curves. See curves, equal loudness. Signal amplitude or sound intensity level (SIL) measured in decibels (dB), and relative loudness levels (LL) indicated in Phons, are obviously not one-to-one correlates for human ears, especially across all frequencies, and particularly not so at lower intensity levels. • Now, a bit of informed speculation: to the discerning musical ear, a sine wave audio signal with a fixed amplitude also has a different timbre in different frequency registers. Who could equate the tonal quality of a keening 4 kHz sine wave with the flaccid, tubby tone quality of a 200 Hz sine wave, given the same signal amplitude? Obviously, the ear plays a central role in perception of timbre. This observation does not conform to (badly informed) "conventional wisdom," where waveform and timbre are often equated unequivocally, but this change of timbre where there is absolutely no change of waveform or amplitude seems quite apparent. Again, perception of timbre is highly subjective. Waveform, line spectrum, and signal amplitude are objective realities. To summarize, radically different waveforms might have the same perceived timbre, due to our inability to discriminate phase or polarity changes of constituent partials. Conversely, the same waveform (e.g. sine waveform on equal loudness curves) evidently can have different timbres as well. (The latter statement is admittedly highly subjective, but one that both authors perceive). Finally, overall loudness level has a bearing on our perception of timbre, as well as other factors discussed, due to considerations such as the nonlinear response of the human ear at high intensity levels. See spectrum. See waveform. See formant. See loudness curves, equal. See correlate.

time domain • See domain, time. Cf. domain, frequency.

toggle • (verb) To change from one state, value, signal, or status of a pair to the alternate state, value, signal, or status. • (noun) A switch that enables changes between a pair of alternate states, etc. See switch, toggle.

tonal balance • The relative levels, or amplitudes of partials in various frequency bands in a complex waveform or in an audio recording mix. Particularly, the relative strengths of low, mid, and high frequency bands, especially in an audio signal or audio mix, represent tonal balance. When partials are boosted or attenuated disproportionately (e.g. by filtering or equalization), the tonal balance is altered. For human hearing, a significant alteration of the tonal balance of an audio signal is perceived as changing, or "coloring" its timbre (tone color). See timbre. See EQ.

touch sensitivity • Ability of a musical interface device, particularly a keyboard, to respond to varying degrees of force applied to its key(s), or to the velocity (speed) at which keys are depressed and (in some designs) released. See after touch. See velocity. See force.

track • A subdivision of hardware (e.g. analog magnetic audio tape) or software (MIDI sequencer) system memory or recording elements that allow differentiation from other such elements. A track typically contains data, e.g. MIDI notes, or audio sound files or recordings, that are kept distinct from other tracks, in order to facilitate independent treatment or processing of data on such tracks.

transducer • Device that changes one form of energy into another, ideally without distorting or losing information that might be transmitted or carried by the original form of energy. For example, a microphone, or mic (rhymes with "like") transduces kinetic energy, e.g. sound pressure waves caused by physical vibrations, into potential energy, e.g. fluctuations of voltage (electrical energy). A loudspeaker reverses this process.

transfer function • Mathematical relationship, graphed as a curve or line, that describes the output of a device based on the digital or analog procedure, operation, mathematical process, or algorithm that treats, transforms, or processes the input signal.

transients, attack • The word "transient" is defined generally as an event or thing that occupies a short time, quickly disappears, or resides in one place briefly. • Attack transients are the highly dynamic amplitude, frequency, and (possibly) phase changes that partials in a complex wave undergo during the beginning of an articulated sound. Attack transients are followed by the less dynamic sustained part of the sound called the "steady state." Cf. transients, decay. See partial. • When both attack and decay (aka release) transients are removed from solo recordings of orchestral instruments that sound the same pitch, it becomes more difficult to distinguish among those instruments when listening solely to their residual steady states, i.e. static timbres, or tone colors. Transients provide critical cues that facilitate perception of vocal or instrumental timbre, i.e. tone color. See timbre.

transients, decay • The word "transient" is defined generally as an event or thing that occupies a short time, quickly disappears, or resides in one place briefly. • Decay (aka release) transients are the highly dynamic amplitude, frequency, and (possibly) phase changes that partials in a complex wave undergo during the ending of an articulated sound. Decay transients are preceded by the less dynamic sustained part of the sound called the “steady state.” Cf. transients, attack. See partial. • When both attack and decay (aka release) transients are removed from solo recordings of orchestral instruments that sound the same pitch, it may become difficult to distinguish among those instruments when listening solely to their residual steady states, i.e. static timbres, or tone colors. Transients provide critical cues that facilitate perception of vocal or instrumental timbre, i.e. tone color. See timbre.

transverse • Situated across or proceeding in a crosswise direction, as in a transversally mounted automobile engine (at a right angle, or 90° (ninety degrees, or 1/4th of a circle) to the axis of the fixed wheels of the vehicle not used for steering). • In the physics of wave propagation (travel), a displacement of energy at a right angle (90°) to the direction in which a wave is propagating. The wave (vibration) on a violin string is a transverse wave. That is, a “kink” of energy flows at 90° to the axis parallel to the length of the string, reflecting repeatedly from one end of the string to the other. The period (time) of this excursion gives rise to the frequency of the vibration produced, and hence the pitch we perceive. See period. Cf. longitudinal.

tremolo • A form of amplitude modulation (AM) that produces slow (2–7 Hz), smooth (quasi-sinusoidal), repetitive (quasi-periodic), shallow (low amplitude) changes of the loudness (amplitude) of an audio signal. Amplitude modulation (AM) may be applied to any signal connected to the signal input of a properly biased two quadrant multiplier (SCA) or any four quadrant multiplier, using a control signal connected to the control input of the multiplier. See SCO/SCF/SCA. See VCA. See amplifier, voltage controlled. See multiplier, two quadrant. Cf. multiplier, four quadrant. In math terms, tremolo may result from multiplication of the carrier (signal input) and modulator (control input) signals connected to the SCA. The low frequency (2-7 Hz) modulator signal typically determines the rate of tremolo. Cf. vibrato. • While aurally similar to tremolo, beats are caused by interaction(s) of two or more periodic audio signals (pitched sounds) that fall within the span of human hearing, e.g. dual strings on a 12 string guitar or multiple strings for a single note in the middle register of a piano. Beats occur when at least two audio signals are summed algebraically (±), which can produce repetitive low frequency loudness pulsations that sound somewhat like tremolo. In math terms, beats result from superposition, or addition of periodic signals whose frequencies are only slightly different, resulting in patterns of interference that are alternately constructive (louder), then destructive (softer). In contradistinction, tremolo is produced by multiplication of at least two signals whose frequencies are typically widely different. • Tremolo is the primary component of the mechanically produced tremulant feature of theater organs such as the “Mighty Wurlitzer” (pipe organ) used to accompany so-called “silent” (meaning narration-free and dialogue-free) films prior to the popular advent of the sound film (ca. 1927). This tremulant feature on theater pipe organs has been described erroneously as “vibrato,” which is a deviation of pitch, not loudness. Certain brands of guitar amplifiers have also erroneously been described as producing “vibrato,” when the effect is actually tremolo. The “Leslie” speaker effect produced in conjunction with a Hammond Organ is due to (periodic) Doppler effect, and really does involve changes of frequency, and therefore is a form of “vibrato,” albeit with some tremolo as well. See Doppler effect. Cf. vibrato.

trigger signal • A bi-state step signal that can be made to briefly change from one of its levels to the other, after which it quickly returns to its original level. In contrast, a gate signal remains at one level indefinitely until it is caused to produce its alternative level, at which it then remains indefinitely, and so forth. A trigger is a step signal characterized by a brief signal pulse. A gate is a step signal that is latched until its alternative state, or level is initiated. See signal, step. Cf. gate.

trigger input • Input designed to respond to a trigger signal. See trigger signal. Typically an input that responds as a threshold detector. See trigger. See input, timing. Cf. gate input. See threshold detector.

truth table • Diagram that shows all combinatorial possibilities for inputting zeroes (0) and/or ones (1) into a given logic gate input, and the resulting logic gate output of either zero (0) or one (1). In terms of "truth," a one (1) is taken to be "true," and a zero (0) is "false." See logic gate.

tweeter • Speaker or driver in an audio monitor, public address (PA), or loudspeaker system specifically designed to accurately reproduce the highest frequencies in a music or sound program transduced and broadcast by the system. • A device that transduces electrical current into sound.

ultrasonic • Frequencies higher than the upper limit of human hearing (20 kHz). Not to be confused with the term "supersonic," whose use is inappropriate in the sonic arts. Cf. supersonic.

unbalanced line or cable • See cable, unbalanced. Cf. cable, balanced.

unipolar • See (polarity) unipolar. Cf. (polarity) bipolar.

unit generator • Module in the Music N series of direct digital synthesis software programs pioneered by Max V. Mathews and associates at Bell Labs dating from the late 1950s. Unit generators create, process, and control sound in a direct digital synthesis environment, i.e. within the confines of a digital computer. Unit generator is the generic name for modules from these first virtual synthesis systems.

unity gain • Representation of the status or performance of a signal processor that reflects neither amplification nor attenuation of the processed signal. A gain, or factor of multiplication that yields a ratio of 1:1 between output and input levels. See amplifier, unity gain.

variable • Mathematical symbol or letter such as "x" or "y" that can be given different values arbitrarily, i.e. at will. A variable is often a temporary "holder" of a mathematical value that is intended to change or be updated due to subsequent calculations. Cf. constant. • A constant such as the speed of sound (v) may change, e.g. due to the temperature, density, and/or elasticity of the medium through which sound propagates.

However, the value of a constant currently in operation must be maintained during execution of a specified set of mathematical calculations or scientific experiments to qualify as a constant. Many constants are derived due to laws of physics; e.g. the speed of sound is constrained to fall within a particular span of possible velocities, depending on a variety of factors (see above). In contradistinction, variables may be given any value desired arbitrarily (at will).

VCA • Voltage controlled amplifier (VCA), terminology for a processor (two quadrant multiplier) dating (in the field of modular synthesis) from the introduction of analog voltage controlled synthesizer modules circa 1964. A VCA alters the amplitude of the (carrier) signal passing through it, proportional to the sum of the instantaneous levels of the signals connected to its internal (bias) and external (modulation) control inputs. • Virtual implementations of modular systems may continue to use terminology such as "VCA," even though such references to voltage (V) control are vestigial (solely of historical interest; no longer relevant) in a software system. See SCO/SCF/SCA. See multiplier, two quadrant. Cf. multiplier, four quadrant.

VCF • Voltage controlled filter (VCF), terminology for a processor dating from shortly after the introduction of analog voltage controlled synthesizer modules in 1964. The cutoff frequency of a VCF is changed, proportional to the sum of instantaneous levels of the signals connected to its internal (bias) and external control inputs. • Virtual implementations may continue to use terminology such as "VCF," even though such references to voltage (V) control are vestigial (solely of historical interest) in a software system. See SCO/SCF/SCA.

VCO • Voltage controlled oscillator (VCO), terminology for a wide range, periodic signal generator dating from the introduction of analog voltage controlled synthesizer modules in 1964. The frequency of a VCO is proportional to the sum of the instantaneous signal levels connected to its internal (bias) and external control inputs. • Virtual implementations may continue to use terminology such as "VCO," even though such references to voltage (V) control are vestigial (solely of historical interest) in a software system. See SCO/SCF/SCA.

velocity • Rate of change of the displacement of a body that moves in time. Velocity is illustrated by a vector (arrow-like graphic) that indicates both magnitude and direction. • Velocity and speed are not precisely equivalent terms. Speed is defined as the rate of change of distance traveled in time in a straight line or a curve that is constant. When these terms are interchanged casually, it does no violence to an artist's work, but the technical distinction remains valid.

verbal • Using words, or having to do with words, whether spoken or written. Cf. oral. "Verbal" is often used incorrectly, e.g. when "oral" is the intended meaning, as in a "verbal" agreement–which is correctly an oral agreement.

vernier • Slider or knob with an auxiliary part or mechanism that moves in concert with, but with greater resolution than the coarse control provided by the central part or mechanism.

vibration • Oscillations of physical elements of a fluid or an elastic solid, or of an electromagnetic wave. The term is usually applied to oscillations that can be perceived by one or more of human senses (e.g. hearing, touch, vision).

vibrato • Frequency modulation (FM) characterized by slow (2–7 Hz), smooth (quasi-sinusoidal), repetitive (quasi-periodic), shallow (low amplitude) changes of an audible audio signal, perceived primarily as smoothly alternating changes of pitch. • Vibrato performed on an acoustic instrument typically introduces sonic byproducts that accompany the simple pitch deviations produced. For example, the body of a violin functions like a comb filter, with a frequency response that features many amplitude peaks (boosts) and valleys (cuts). Violin vibrato is produced by altering the length of the stopped portion of a string by rocking the stopping (fretboard) finger slowly (2–7 Hz) back and forth in the same axis as the string, particularly while bowing the string. (Vibrato may also be applied briefly to a plucked string.) The frequencies of all partials (mostly harmonics) produced by the violin string consequently move higher and lower proportionately during vibrato as well. The violin body's fixed comb filter response causes dynamic amplitude changes of these harmonics individually as overall string frequency is altered. Violin vibrato therefore causes significant dynamic audio waveform, i.e. spectral (timbral) changes, as well as the obvious changes of pitch. In instruments such as the flute, vibrato may be accompanied by a near-synchronous tremolo, or slow amplitude modulation (AM) at the vibrato rate, causing changes of loudness as well as pitch. Cf. tremolo. • The distinction between vibrato, a repetitive change of an audible audio signal frequency perceived as pitch, and tremolo, a repetitive change of audible audio signal amplitude perceived as loudness, has unfortunately been muddled due to the use of the term "vibrato" by a guitar amplifier manufacturer to (incorrectly) describe the tremolo capability of some of its products. Cf. tremolo.

virtual • Something that exists in effect rather than in reality, as per any ordinary understanding of the term reality. • Having no physical existence or substance, but simulated or made to appear real through the agency of a digital computer, for convenience, economy, artistic purpose, or to enhance performance or appearance. Computer graphic images (CGI) in films are a good example of such "reality" that is only apparent, not actual. • A description, e.g. virtual memory that describes a technique used in computers to provide "impermanent" storage, or a capacity for storage that appears to be beyond the apparent capacity of permanent storage (memory) provided by the computer.

virtual reality • An oxymoron, or expression of speech that includes words or roots having mutually contradictory meanings. The word oxymoron has roots oxy meaning sharp, and moros meaning dull (from which we derive the word "moron," moron). Reality is apparently rooted in politics, economics, and especially physics. When there are no potatoes to eat, a virtual representation of a potato will prove to provide negligible sustenance–no matter how "real" it might seem. Virtual displays, sound synthesis software, and associated virtual modules have great utility, but the notion of "virtual reality" depends solely upon quixotic reflexive suppositions. When there really are no potatoes to eat, couch potatoes sitting before video display terminals (VDT), and those wearing "virtual reality" helmets looking for a meal will truly be in dire straits (with apologies to Mark Knopfler). Seek your reality elsewhere, grasshopper. Use your ears–not your eyes. Semicircular canals inside one's head play a primary role in our sense of balance and are smarter than our foolish eyes. If your eyes fly around, but your ears don't–it can make you vomit! And, while you are at it, produce some real music, defined to me by Bob Moog as " . . . music that earns some real money."

vocoder, channel • Processor that impresses the dynamic spectral (timbral) characteristics of a program (P) signal aka the modulation (M) signal, onto the typically static spectrum of a carrier (C) signal. The channel vocoder (CV) was developed by Homer Dudley in the mid 1930's, using analog signal analysis and data reduction techniques developed at Bell Labs, with the intention of reducing bandwidth requirements for telephone conversations. The project was superseded at Bell by the advent of PCM (pulse code modulation), a time division multiplexing technique (rather than frequency multiplexing a la Dudley's channel vocoder). Dudley then took his device to Germany, where it stimulated the interest of Bonn phonetician Werner Meyer-Eppler, who participated in the creation of the NWDR (radio station) Köln (Cologne) electronic music studio, celebrated by many as the "first" such facility. • A channel vocoder (CV) has two identical band pass filter (BPF) banks in separate analysis (program signal) and synthesis (carrier signal) sections. Each BPF in the analysis section is routed through an associated envelope follower (EF). Each BPF in the synthesis section is routed through an associated voltage controlled amplifier (VCA). The (carrier) signal outputs of all synthesis section VCAs are summed and monitored. A BPF and envelope follower (EF) in the analysis section, with its associated BPF and voltage controlled amplifier (VCA) in the synthesis section, constitute a channel. Per each channel, EF output from the analysis portion of a channel is normally connected to the VCA control input in the synthesis portion of the same channel. Each channel's envelope follower (EF) outputs a slow moving envelope signal that represents the average signal power in its associated analysis section BPF from moment to moment. Typically, the two band pass filters in each channel have the same fixed center frequency (f_{ctr}). Therefore, spectral changes reflected by EF outputs in the analysis section are impressed on whatever signal is connected to the synthesis section. That is, the envelopes produced in the analysis section open and close associated VCAs in the synthesis section. For example, when an audio frequency pulse waveform is connected to the carrier (C) input, or synthesis section of a CV, the vocoder "talks" as one speaks into a microphone connected to the program (P) input, or analysis section. Pitch is then based on the frequency of the pulse wave (carrier), which functions like human vocal cords. The microphone (program) impresses its dynamic spectral (timbral) changes onto the carrier, just as the tongue and oral cavity shape the crude utterances of the human voice box. Typically, a channel vocoder (CV) has multiple band pass filters with frequency bands that overlap to encompass much of the audio frequency bandwidth. A channel vocoder (CV) is based on a dual set of filter banks that process program and carrier signals essentially simultaneously. In some designs, the frequency of the program signal is "followed" from moment to moment, and these data may be used to make the carrier signal (oscillator) re-create these frequencies. • In some CV implementations, the normaled connections from EF outputs to VCA control inputs in a channel can be "scrambled" among channels. In this case, the output of a given analysis channel EF might be connected to the control input of a synthesis channel VCA that has a different BPF center frequency. The center frequencies of the two band pass filters in such a scrambled channel configuration likely will not have the same fixed center frequency. High frequency bands in the program signal (voice) might control low frequency bands in the carrier signal (pulse waveform), or EF output to VCA control input connections might be assigned randomly. Creative results that are only vaguely speech-like may result. See envelope follower. See VCA. Cf. vocoder, phase.

vocoder, phase •

volt • Electromotive force (EMF), or the "pressure" of electricity in an electrical circuit, by analogy with water pressure. A difference in voltage, aka a potential difference between two points in an electronic circuit or electrical grid can cause electricity to flow. • The SI unit of 1 volt is the amount of electromotive force (EMF) required to move an electric current of 1 ampere through a resistance of 1 ohm. • The volt is named after Count Allessandro Volta (1745–1827), the illustrious Italian physicist who built the first electric battery known as the Voltaic pile.

voltage control • System of controlling module parameters where signal characteristics are altered proportional to the sum of applied analog control signal(s) comprising voltages, in the appropriate voltage range(s) dictated by the particular design. In the most popular design(s) for early (1964) analog music synthesizers, a 1 volt per octave (1volt/octave) relationship evolved as a standard.

volume • Relates to judgments about the perceived size of a sound, in contradistinction to its perceived loudness. Both volume and loudness are constructs based on human perception, and such subjective judgments are inherently difficult to quantify directly. Also, unfortunate common usage has made the terms "volume" and "loudness" appear to be practically synonymous. However, perception of volume may depend more on particular aural cues (e.g. time of the first return, i.e. the first echo of reverberation), that suggest the size of the space in which a sound apparently resides. Also, sounds created by many point sources of sound, a multiplicity of musical voices sounding simultaneously, (e.g. a symphony orchestra string section), may also affect perception of size, which may equate more to the construct of volume than to loudness. • Judgments about loudness are apparently based primarily on perception of sound intensit(ies) at various frequencies, which can be reported in relative loudness levels (LL) quantified in Phons (pronounced like the "phones" of "telephones"). Phons can in turn be related to objectively measured decibel (dB) power levels. Perception of loudness is quantified indirectly using relationships between Phons and decibels. (Decibel: an objective measurement of the ratio of two intensities, or powers). See equal loudness curves. Our perception of volume is elusive, and even more difficult to quantify than loudness.

Attention graduate students! There may be a dissertation topic (or two) here. Cf. loudness.

vs. or v. • Abbreviation of versus, meaning opposed to, compared to, or contrasted with.

VU meter • A volume unit (VU) meter is typically used to display the power of non-sinusoidal electronic signals, particularly those of interest to an audio engineer, e.g. music, speech, and sound effects (SFX). Zero (0) VU level may be arbitrarily calibrated as the level equivalent to one (1) milliwatt, i.e. the power produced by a sine wave of a particular level routed through an impedance of 600 ohms. That is, zero (0) VU may be calibrated to be equivalent to the zero (0) decibel reference 0 dBm, which represents a level of approximately 0.775 volts (across 600 watts). (The small "m" in "dBm" stands for milliwatt.) See ohm. See impedance.

Walsh functions • Group of rectangular (binary-state) waves used to approximate or constitute a variety of other periodic signals having various waveforms. Walsh functions can take on only two values: –1 or +1, but their transform(s) can produce signal(s) having many possible values. • In general terms, this is a form of additive synthesis, defined as a less frequently used technique that uses simpler waves as the basis of construction of more-complex waves. Cf. synthesis, additive.

watt • SI derived unit of measurement for electrical power, expressed by the formula P = I E where power (P) in watts equals current (I) in amperes times voltage (E) in volts. • In other terms, 1 watt (W) is equal to 1/746 horsepower or 1 ampere at 1 volt. • The watt is named after James Watt, the British engineer who originated the term "horsepower," and pioneered construction of steam engines and propellers for marine vessels.

wave • Movement of a physical entity to and fro, back and forth, up and down. Or a signal characteristic exhibiting or causing such alternating changes of voltage, current, power, pressure, or energy. A wave is generally considered to be periodic, but aperiodic pulses may also be considered to be waves. Waves can propagate (travel) through a material elastic medium such as air, e.g. sound waves. See wave, longitudinal. Cf. wave, transverse.

wave, longitudinal • Wave whose direction of propagation and changes of energy level, pressure, amplitude, etc. are aligned on the same axis. Sound waves are longitudinal waves. Cf. wave, transverse. See transverse.

wave, transverse • Wave whose direction of propagation and changes of energy level, vibrations, amplitude, etc. are aligned at a right angle (90°). Vibrations, i.e. the waves on a violin string are transverse waves. However, the sound waves produced by the violin string and body of the instrument are longitudinal waves. Cf. wave, longitudinal. • Longitudinal (sound) waves are depicted as transverse waves by an oscilloscope, thereby enhancing our ability to measure and interpret sound waves. See transverse. Cf. longitudinal.

(wave propagation) absorption • The loss of acoustic energy, caused primarily by a change of sound energy into heat. Absorption occurs when a sound wave falls onto or propagates through any material that can absorb. The formula for amount of absorption, represented by the Greek letter "alpha" (α) is: α = absorbed energy/total incident energy. That is, the resulting alpha (α) coefficient represents the ratio of the amount of energy absorbed, relative to the "incident," or total energy that falls upon the surface of interest. The alpha coefficient therefore ranges between zero (0), i.e. no absorption, and one (1.00), i.e. complete absorption, although such theoretical limits are highly unlikely in the real world. Alpha coefficients are published for frequencies at octave, ½ octave, 1/3 octave, etc. (frequency) spans for specific materials such as glass, draperies, concrete, ceiling tiles, etc. to assist in the design of lecture halls, recording studios, and performance venues.

(wave propagation) diffraction • The behavior of a wave as it bends, or spreads as it passes around the edge of an obstacle or through a relatively narrow aperture (opening). The bell of a brass instrument is an opening through which a complex wave having partials at many frequencies can pass. The size of the bell (aperture), can be measured as though it were a wavelength, from which an equivalent "boundary" frequency can be computed by: f = v / □ where "f" is frequency, "v" is the velocity of sound (344 meters/second @ 20° Celsius), and "□" is lambda (the wavelength analogous to the diameter of the bell). Partials in the complex wave at frequencies higher than this boundary frequency "beam" essentially straight through the bell, and those lower than the boundary frequency "bend" (diffract) as they pass through the bell. This partially explains why a brass instrument sounds "brighter" when its bell is pointed directly toward a listener (or a microphone). Higher partials in the complex wave passing through the bell "beam" along the axis in which the bell is pointed, and lower partials "bend" (diffract) away from this axis. Brass instruments are "directional" because the ratio of "beamed" to "diffracted" energy is larger as the instrument is pointed directly toward the location of a listener or mic.

(wave propagation) interference • Algebraic (±) summing that occurs when two or more propagating waves intersect. Two intersecting sine waves with the same frequency illustrate constructive interference when the pressure at their intersection, with respect to ambient barometric pressure, is greater than the largest pressure either of those waves could cause individually at that point. Decreased pressure at such an intersection is due to destructive interference. • Reflections of sound waves in a room cause louder or quieter locations in the room for each frequency, due respectively to constructive or destructive interference. See (measurement unit) decibel. See wavelength. • Two periodic waves with frequencies that are close, but not identical, create repetitive loudness pulses called "beats" as a result of wave interference that is alternately constructive (louder), then destructive (quieter), etc. See beats. Cf. tremolo.

(wave propagation) reflection • Phenomenon that occurs when a sound wave meets a sudden change of acoustical impedance (resistance) in the varying media through which it propagates (travels). Echoes in stone canyons are a common sense example, where a sudden increase of acoustical impedance (stone compared to air) causes such reflection(s). However, the requisite sudden and significant change of acoustical impedance does not have to be an increase, as in this example. Any sudden decrease of acoustical impedance also causes reflection(s). For example, in brass instruments, the pulses (waves) produced at the mouthpiece propagate through the tubing of the instrument, and then encounter a sudden decrease of acoustical impedance a short distance after leaving the bell of the instrument. This sudden decrease of acoustical impedance causes those pulse(s) to reflect back into the bell of the instrument, setting up resonant modes within the horn that create a harmonic series that facilitates the possibility of playing an intelligible musical scale. (Modern trumpets use valves that change the length of the tube, making chromatic music possible). The (incomplete) harmonic series on a bugle, for instance, exemplifies the resonant modes produced due to such reflection(s) back into the instrument. Like the trumpet, a bugle is missing the first harmonic (H1) of an otherwise complete harmonic series. The lowest "open" note is H2, as revealed by the interval of a perfect fifth (P5) to the next higher harmonic (H3), aka the third harmonic.)

(wave propagation) refraction • Generally, a change of direction of a wave when it passes from one medium to another medium with a different density. • Bending of a sound wave due to ambient conditions that typically constitute a gradient (measurement of a gradual change over a distance), e.g. of densities in the medium through which the sound wave propagates (travels), e.g. horizontal layers of air having different temperatures.

waveform • Shape of the vibration or representation of vibration or wave movement in two alternating directions or signal polarities, as depicted in time (t). That is, alternating conditions of a physical parameter such as voltage, current, power, etc. or its virtual equivalent depicted in time (t). • Such bidirectional (±) movement can be displayed as a bipolar signal (±) in the time domain. See domain, time.

waveform, aperiodic • Aperiodic means "not periodic." A random signal (e.g. white or pink noise) is aperiodic. That is, unlike a periodic signal produced by an oscillator, an aperiodic signal produced by a noise source has no specific cycle that completes during a specific period (T) of time (t), nor a cycle that repeats endlessly during such ensuing periods. This is an essential requirement for randomness. See waveform, random. However, a non-random signal can also be aperiodic (e.g. envelope generator (EG) output signal. See ADSR.) That is, all random signals are aperiodic, but not all aperiodic signals are random. Cf. waveform, periodic.

waveform, complex • Signal consisting of the algebraic (±) summation of more than one partial (sine or cosine wave). See waveform. Cf. waveform, sine.

waveform, geometric • Periodic waveform constituted by trigonometric functions such as sin (sine) and cos (cosine). • Sawtooth, pulse (square), triangle, and sine waveforms are typical geometric waveforms produced by a periodic function generator, aka an oscillator. See waveform, pulse etc.

waveform, periodic • Signal with a specific cycle that completes during a finite interval of time, or period (T), and repeats endlessly per each identical period that follows. From a purely theoretical standpoint, any periodic wave has always existed, and will exist throughout all time! • A waveform with a cycle that replicates, or repeats a specific number of times in the standardized time of one second. The number of cycles in one second is the frequency of a periodic waveform. Frequency (f) has an inverse relationship with period (T), that is: $f = 1/T$ and conversely $T = 1/f$. A periodic signal has a specific frequency that, when located in the frequency span of human hearing, can create a sound with a definite musical pitch. See frequency. Cf. pitch.

waveform, pulse • Geometric waveform, as depicted in the time domain, that has only two levels, which alternate in time periodically. One complete up-down change of level constitutes one cycle (period) of the pulse waveform. • The percentage of time the high state is present (remains on) during the period (T) of a pulse wave represents its duty cycle. For example, a square wave is a particular kind of pulse wave that has a duty cycle of 50%, i.e. the square wave's "high" state occupies 50% of its period (at any frequency). Harmonics with even numbers H2, H4, H6, H8, etc., are missing in a square wave, leaving only odd harmonics H1, H3, H5, H7, H9, etc. The square wave's 50% duty cycle is equivalent to the fraction 1/2. The denominator ("2") represents the number of the initial harmonic of a group of missing harmonics with whole numbered harmonics at multiples of this denominator also missing. That is, for the square wave, harmonics at multiples of "2" are missing (2, 4, 6, 8, 10, etc.) Any duty cycle that can be reduced to such a simple fraction made up of whole numbers points to the spectrum of that particular duty cycle of a pulse wave. For example, a 10% duty cycle is equivalent to the fraction 1/10, and such a pulse waveform consequently has harmonics missing at H10, H20, H30, H40, etc. (every tenth harmonic starting with H10). A 25% duty cycle pulse waveform (1/4) lacks harmonics H4, H8, H12, H16, etc. (every fourth harmonic starting with H4). The amplitude(s) of harmonics present between such missing harmonics become progressively smaller as they approach the frequenc(ies) of missing harmonics on either side. That is, many pulse waveforms have a line spectrum with harmonics that feature characteristic rising and falling amplitude "scallops" between missing harmonics. See spectrum, line. See waveform duty cycle. See waveform, geometric. See harmonic. See your shrink.

waveform, ramp • Alternate name for a sawtooth waveform, particularly one that starts at zero (0), rises to its maximum level using an oblique (slanted) straight line, and returns to zero (0) (ideally) immediately at the end of its period (T), and then restarts and repeats this cycle.

waveform, random • Aperiodic waveform with partials exhibiting unpredictable (random) amplitude and phase fluctuations, in the case of white or pink noise. See noise, etc.

waveform, rectangular • Alternate name for a pulse waveform. See waveform, pulse.

waveform, sawtooth • Geometric waveform with all harmonics, indicated as H1, H2, H3, H4, H5, etc., based on any fundamental frequency, aka the first harmonic, or H1. Given a fundamental (H1) of 100 Hz, a sawtooth waveform has harmonics at 100 Hz, 200 Hz, 300 Hz, 400 Hz, 500 Hz, etc. • In the sawtooth waveform, any harmonic's amplitude "a" is the reciprocal of its harmonic number "n," that is: ($a = 1 / n$). For example, the amplitude of the fifth (5th) harmonic (H5) of a sawtooth wave is 1/5th that of the amplitude of its fundamental (H1), or first (1st) harmonic. See waveform, geometric. See harmonic. See relationship, reciprocal. See coefficient.

waveform, sine • (Pronounced like stop "sign.") A waveform consisting of a single partial, having no associated harmonics or inharmonics. The Latin word sine (pronounced see´nay) means without, meaning in this context, without any other partial(s). A single cycle of this sine waveform represents 360° of circular motion, which is equivalent to 2π radians. Circular motion can be illustrated by the projection in time of a rotating vector. This allows depiction of the waveform in the time domain. See domain, time. Each rotation from zero degrees (0 °) through three hundred sixty degrees (360 °), or once around the circle, constitutes one period (T) of the waveform. The sine function also illustrates simple harmonic motion (SHM), e.g. a simple pendulum, and also circular motion of a sin function in trigonometry, therefore a geometric waveform depicted over time. The sine waveform is symmetrical about both (x) and (y) axes in a time domain representation. See domain, time. • The sine wave, due to its uniqueness, and cheerful acceptance of the self-imposed solitude of its single partial, is aka the "Irish wave: 'tis itself!"

waveform, square • Geometric pulse waveform with a 50% duty cycle. See waveform, duty cycle. • Geometric waveform with odd number harmonics H1, H3, H5, H7, H9, etc. Given a fundamental (H1) of 100 Hz, a square waveform has harmonics at 100 Hz, 300 Hz, 500 Hz, 700 Hz, 900 Hz, etc. • In the square wave, any harmonic's amplitude "a" is the reciprocal of its harmonic number "n," that is: ($a = 1 / n$). The amplitude of the fifth harmonic (H5) of a square waveform, for any fundamental frequency is ($a = 1 / 5$), or 1/5th of the amplitude of that fundamental frequency (H1). See waveform, geometric. See harmonic. See relationship, reciprocal.

waveform, triangle • Geometric waveform with odd number harmonics H1, H3, H5, H7, H9, etc. Given a fundamental (H1) of 100 Hz, a triangle waveform has harmonics at 100 Hz, 300 Hz, 500 Hz, 700 Hz, 900 Hz, etc. • In the triangle waveform, any harmonic's amplitude "a" is the reciprocal squared of its harmonic number "n," that is: ($a = 1 / n^2$). Therefore, the amplitude of the fifth harmonic (H5) of a triangle wave, at any fundamental frequency is ($a = 1 / 5^2$), or 1/25th of the amplitude of that fundamental frequency (H1). See waveform, geometric. See harmonic. See relationship, reciprocal squared. • A triangle wave constructed using coefficients reveals that successive harmonics alternate between positive and negative (±) polarities. A triangle wave with a fundamental (H1) of 100 Hz has harmonics at +100 Hz, –300 Hz, +500 Hz, –700 Hz, +900 Hz, etc. These alternate polarity changes are rarely depicted in a line spectrum that purportedly accurately illustrates the harmonic makeup of the triangle waveform. Typically, a simplified line spectrum shows harmonics with the same (+) polarity, implicitly justified by the ear's relative insensitivity to the polarity of partials within any complex waveform. Nevertheless, the waveform produced by such a simplified (+ only) line spectrum does not produce the familiar triangle waveform when depicted in the time domain. See coefficient.

waveform duty cycle • The percentage of time a pulse waveform (e.g. square waveform) maintains its higher signal state during its period (T). The square wave, a type of pulse wave, has a 50% duty cycle. See waveform, pulse. • Most periodic function generators (oscillators) that produce pulse waveforms are designed to maintain the selected duty cycle at any frequency produced, thereby maintaining a waveform with essentially the same timbre over the entire span of frequencies generated.

wavelength • In acoustics, the distance a periodic sound wave travels through a material elastic medium such as air during that wave's period (T), i.e. the time (t) of one complete three hundred sixty degree (360 °) cycle of its waveform. Wavelength is aka lambda, represented by the Greek letter (□). • The velocity (v) of sound in air is 344 m/s (meters per second) @ 20 ° C, aka "room temperature" (20 ° Celsius or Centigrade). The formula that relates velocity (v), frequency (f), and lambda (□) is: v = f □ as recalled by the mnemonic "very fine liquor," (where velocity (v) is "v" for "very," frequency is "fine," and lambda, or wavelength is "liquor.") Solving for lambda (□ = v / f) shows that 1000 Hz has a wavelength (□) of 0.344 meters, the approximate length of an American foot long hot dog (frankfurter). A proper Coney Island "foot long" hot dog should be at least thirteen inches (13") long, in the same way that a "baker's dozen" is thirteen (13). • Wavelength is shown as the distance from a selected point on a periodic wave to the corresponding point on the wave in the ensuing (next) period, typically from crest to crest, as the wave would appear in the time domain. • White noise and pink noise have no definable wavelength (□), because a random or aperiodic signal has no discernible cycle that repeats. Having no period equates to having no measurable wavelength. Nevertheless, both aperiodic and periodic sound waves propagate through air at a velocity of 344 m/s @ 20° Celsius.

waveshaper, nonlinear • Processor purposely designed to distort input signal(s), in order to produce partial(s) not originally present in the input signal. • Processors and other devices can be forced to distort signals by driving them beyond their nominal design limits, e.g. amplifier clipping, analog audio tape saturation. But a nonlinear waveshaper is specifically designed to exploit a (typically) programmable nonlinear transfer function that may also graphically depict output as a function of input. Some implementations provide for alteration of the function by manipulating its graphical representation, i.e. by directly drawing the curve. See transfer function. • Nonlinear waveshaper(s) are an important part of virtual synthesis engines, often available as effects (FX) processors following the output of the actual synthesis engine (e.g. subtractive, additive, linear FM, physical modeling).

wave table • An array of numbers, or a data file in a digital computer that can be converted into an audio signal by a digital to analog converter (DAC). See converter, DAC. • Typically, a wave table is a digitized audio file in computer memory that is repeatedly scanned. Array elements are accessed, or read serially and repeatedly in order to produce sound. This process is aka as table-lookup synthesis.

woofer • Component, or individual driver in an audio monitor, public address (PA), or broadband loudspeaker system specifically designed to accurately produce low frequencies in music or sound produced by the system. • The very low frequencies (< 100 Hz), are often produced by a subwoofer loudspeaker housed in a separate cabinet.

word, data • Byte, or grouping of bits in a computer or digital information system that facilitates communication or transmission of information. See bit.

x-y generator • A two output generator that continuously senses physical motions that represent locations in two dimensional space, and outputs a coordinate pair (x,y) of signals or numbers that represent the current location. • A joystick is an unusual "dual" x-y signal generator (or processor) that produces a pair of coordinate pairs, i.e. separate location(s) on two associated Cartesian planes. When a joystick is used as a pan pot, the two Cartesian planes are aligned in the same plane, but at a 90° axis to each other. A joystick can be used to move sound from side to side (L–R), and from front to back, e.g. in a quadraphonic (surround, or 5.1) playback environment. Cf. (x,y) values.

(x,y) values • Coordinate pair (x,y), or two numbers that represent a single point on a two dimensional graph known as a Cartesian plane. The (x) value (abscissa) lies on horizontal axis, and the (y) value (ordinate) lies on the vertical axis. See coordinate pair. See abscissa. See ordinate.

zero • The numerical symbol "0" (zero) positioned between positive (+) and negative (–) numbers on the real number line. Zero is the origin of the real number line. Zero represents the absence of quantity or magnitude. • The sound pressure level (SPL) that causes no alteration of ambient barometric pressure, e.g. when propagating sound waves cross zero (0) level between compressions (+) and rarefactions (–). An unbiased monitored audio signal that is symmetrical about the (y) axis, e.g. square or sine waveforms, causes equivalent compressions (increases), and rarefactions (decreases) of pressure of equal magnitude alternately with respect to its zero signal level. • The reference power, or level (0 dB) against which power(s) are compared using the decibel (dB) scale. In this context, zero (0) dB does not mean zero (0) or no power, or zero (0) or no level. Zero decibel, or 0 dB is a specific reference against which other intensities or powers are compared. For instance, "0 dB" on a volume unit (VU) meter is a relatively large signal (approximately 0.775 volts) near the high end of this meter's scale. The majority of program signals monitored by a VU meter are represented by negative dB levels with respect to this 0 dB reference. See (measurement unit) decibel. See Bel. • Zero (0) is the level or numerical value of a control signal, e.g. keyboard or envelope generator, LFO, etc., that causes no change of the parameter it is controlling (e.g. audio oscillator frequency), regardless of the extent to which that zero (0) control signal is attenuated or amplified. Locating the zero (0) value in a control signal provides means of determining the polarity (unipolar or bipolar) of that signal, information that is not always provided by documentation. When the location of zero (0) level is known, the polarity of a signal may usually be tested by ear (by attenuating or amplifying the control signal and taking note of direction(s) in which the controlled parameter moves, due to the polarity of the controlling signal.

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