WORDS TO THE DIGITALLY WISE

A short glossary of terms that will help guide you on your quest for digital audio quality…

AAC (Advanced Audio Coding) An audio compression codec that is part of the MPEG-2 and MPEG-4 standards, and that offers better resolution than the MP3 codec, which is also part of the MPEG standard. The inclusion of AAC as a file format for Apple's iTunes was big step forward in digital music quality.

A/D Converter (analog-to-digital converter, also expressed as ADC) transforms analog audio, such as music on tape or on vinyl, to various digital file formats. ADCs are either integrated into consumer electronics devices or can be stand-alone units. The quality of a piece of digital music is largely dependent upon the quality of the ADC.

AIFF (Audio Interchange File Format) is one of several file formats for storing digital audio data. AIFF supports a variety of bit resolutions, sample rates and channels of audio.

Algorithm Just when you thought you'd never need high-school algebra again. An algorithm is an equation performed by digital audio devices to process data such as music. It's used in professional audio programs to create reverbs and delays, among other effects.

Codec derives from a jam-up of "code" and "decode," as in a program that encodes data into a particular format, like AAC or MP3 for storage or transfer, then decodes it into a playable form.

D/A Converter (digital-to-analog converter) The opposite of an A/D converter, where audio signals in the form of digital data are reconstructed back into an analog waveform.

dB The decibel (dB) is a unit used to measure the intensity of sound. It's a logarithmic scale, like the Richter Scale used to measure earthquakes, it's particularly useful for Heavy Metal music. We're kidding -- the decibel scale allows us to measure any number of music parameters, from volume to distortion.

Dynamic Range is the range, in dB, between the noise floor of a device and its defined maximum output level.

Equalizer Also expressed as EQ, equalizers are electronic devices that affect the frequency relationships of an audio source. Divided into specific and selectable frequency bands, equalizers allow certain frequencies to be either cut (decreased) or boosted (increased) and measured in dB (that term again). EQ is a tool that can smooth out …

Frequency is the number of cycles that a repetitive waveform makes in one second. For instance, a waveform that repeats once per second has a frequency of 1 Hz (Hertz); a waveform that repeats a thousand times per second has a frequency of 1 kHz (kilo-Hertz). Most consumer audio systems are designed to be able to reproduce sound accurately in a range between 20 Hz and 20 kHz. This range is also referred to as a "frequency response."

Lossy is a method of data-compression encoding that compresses data by discarding (i.e., losing) some of it. Now, before you say, "Lossy, come home!" understand that the algorithms that perform these calculations are looking for sounds whose frequencies are very similar to each other, and they will lose some of that data to make the overall file more compact. When the file is played back, the missing data is, hopefully, not really missed. Lossy formats certainly drove the convenience side of the consumer digital audio equation, but at some sonic cost (though you may never have noticed it over earbuds).

Lossless encoding means just what is seems to: it allows all of the original data to be exactly reconstructed from the compressed data.

MP3 -- The MPEG-1 or MPEG-2 Audio Layer III codec is commonly referred to as MP3, a patented digital-audio encoding format that uses a form of lossy data compression. It has become the de facto standard of digital audio compression for the transfer and playback of music on consumer music devices.

Oversampling In the D/A conversion process, converters sample at significantly higher rates -- at least twice the bandwidth of the signal being sampled. This helps avoid digital distortions such as aliasing (the improper alignment of data with each other), reduces digital noise artifacts, and improves resolution. Think of it this way: if you make soup in a pot only big enough to hold the soup to the brim, it doesn't have a chance to bubble and cook; if you cook it in a bigger pot, you give the soup room to move.

Sample The basic element of a digital image. You know how a movie camera essentially takes a still picture of a running horse 24 times a second, then when you play them in sequence it looks like the horse it running? A sample is a digital photograph, and in audio it's taken 44,100 times (i.e., 44.1 kHz) a second. That's the sampling rate specified for a Compact Disc. Some high-resolution digital audio formats have rates as high as 96 kHz and 192 kHz. The more "pictures" in each one-second slice of sound, the higher the resolution and quality. (The highest frequency that can be usefully recorded, however, is one half the sampling rate, so a CD will reproduce audio up to about 21 kHz, near the upper range of human hearing.)