Digital Audio Recording: Glossary of Terms and Concepts

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A-B

AC Power Adapter: A power adapter for electronic equipment, including field recorders and laptop computers. Plugging the adapter into the wall converts "Alternating Current" or "AC" power to Direct Current or "DC" power. Usually this adapter converts voltages supplied by electrical companies to lower, more regulated voltages that power electronic devices.

AES/EBU: Digital audio standard known as AES3, usually associated with professional level transfer of digital audio in real time. Developed by a collaboration between the Audio Engineering Society (AES) and the European Broadcasting Union (EBU). At this time, AES/EBU is not associated with most portable field recorders but can be utilized in analog to digital converters and sound cards used for digitization.

Analog: In the context of audio, "Analog" refers to the method of representing a sound wave with voltage fluctuations that are "analogous" to the pressure fluctuations of the sound wave. Analog fluctuations are infinitely varying rather than the discrete changes at sample time associated with Digital Recording.

Analog to Digital Converter (ADC, A/D or A to D): Device that converts analog signals to digital. This involves a conversion of the voltages representing the original pressure waves into discrete digital numbers that can be read by computer or digital audio device. A/D converters are found on portable field recorders themselves, as well as on sound cards used for the digitization of analog audio. To achieve archival quality digitization, practitioners must seek out the highest quality analog to digital converters that they can afford. Inferior, consumer based equipment will introduce additional noise into the signal

ASIO: Audio Stream Input/Output is a computer driver for digital audio, originally developed by Steinberg**.** The ASIO protocol is usually associated with higher-end audio hardware and enables the software to directly communicate with the hardware, thus bypassing the audio system built in to the operating system. Native Microsoft Windows audio tends to have very high latency issues whereas Apples OSX does not have the same Latency issues. Users will often have to install the Asio drives while installing their new digital audio interface on the computer. This is not necessary for transferring recordings from a field recorder to a computer, but may be necessary for the computer's audio software to play back the recording utilizing specialized audio hardware interface.

Audio Interchange File Format (AIFF): Digital audio data file format utilizing uncompressed pulse code modulation or PCM. The format was developed by Apple Computer. Most portable field recorders record their uncompressed signals as .wav files. Professional level digital audio editing software packages should include AIFF compatibility however.

Attenuation: A decrease in the level of a signal. A microphone attenuator reduces the output of a microphone and protects the inputs of the recorder from becoming overloaded and therefore distorted. Microphone attenuators appear on recorders, usually with a variety of Decibel settings. Standard attenuators on a field recorder are anywhere from -10db to -30db and allow recorders to utilize microphones that differ in sensitivities and outputs. Example: the Marantz PMD 660's preamps were easily overwhelmed by high-output microphones such as the Audio Technica AT 825.

Audio Editing Software: Computer software designed for capturing and manipulating digital audio. This can involve editing for production, digital signal processing of the audio recording, saving the file in a different format, converting from stereo to mono, compressing a signal, changing the sample rate or bit depth of the data file. Stereo editing software can only handle 2-channels of audio whereas multi-track software can handle multiple channels. Popular editors include:

- Sound Forge (Windows)
- Peak (Mac)
- Audacity
- WaveLab
- ProTools LE

Digital audio editors can be both destructive and non-destructive.:

Automatic Level Control (ALC) /Automatic Gain Control (AGC): A circuit inserted in the preamplification stage of the recorder that quickly adjusts the recording levels to adapt to changes in the audio source signal. This function will work instead of the manual level adjustment capability and will boost a lower end signal as well as protect a hotter signal from distorting. Not usually associated with professional level recording as the recordings tend to be noisy.

Balanced Inputs/Outputs: An protocol for connecting analog audio devices using impedence-balanced cables. This protocol is usually associated with professional level audio equipment and allows for longer cable lengths as well as reducing the addition of external noise to the signal. Balanced cables have either XLR or TRS plugs. Professional level digitization will usually involve balanced outputs on the analog playback device and balanced inputs on the analog to digital converter. A balanced signal can be converted to an unbalanced signal through the use of a DI unit or a transformer. A simple XLR to ¼" adapter will not suffice.

Bandwidth: Refers to the rate audio data is transmitted or the amount of audio data that can be transmitted in a fixed amount of time. It is normally associated with delivering audio over the internet.

Batch Convert: Process of taking any number of files of one type (i.e. wave, mp3, etc.) and changing them into another type. Can involve a change in format or a change in quality settings or from stereo to mono. Moving from type to type is not a two way street however; any changes that compromise the quality of an audio file are permanent.

Bit: A bit is the simplest component in computing. Bits are given a value of either 1 or 0 based on the voltage they are carrying at the time. These bits make up each and every part of the programs and processes that computers encounter with sequences of bits in particular orders define the what, where and why of computing.

Bit Depth refers to the number of bits used to represent a single sample. For example, 16-bit is a common sample size. While 8-bit samples take up less memory (and hard disk space), they are inherently noisier than 16- or 24-bit samples. The higher the bid depth, the better the recording, but also, the larger the file produced. Again, "CD quality" equals 16-bit and should be the minimum bit depth used for field recordings. 24-bit is beginning to catch on in the field as storage media decreases in cost. 24-bit is becoming the standard bit depth for archival quality analog to digital conversion, although many repositories and individuals continue to digitize using 16-bit.

Bit Rate refers to a measurement of digital audio based on the following equation and is usually expressed in kbits/second:

Bit rate = (bit depth) x (sampling rate) x (number of channels)

BNC connector: An alternative to traditional RF coaxial cables, BNC connector cables are mostly used for composite video on commercial video devises. They can easily be connected to a traditional RF jack via an adapter.

"Boomy:" Refers to audio that is accentuating the lower frequencies of a signal too much. Essentially means that the bass signal in a piece of audio is too powerful whether due to the mixing of the actual track being played or the settings on the system it is being played on.

"Bright:" Refers to the clarity of the higher frequencies in a recording. The brighter a piece of audio is, the more clear and distinct these higher frequencies are.

Buffer (Data): The buffer refers to an amount of memory separated from the rest that acts as a holding tank for information as it is taken from one source to another. The changes made to any file are stored in the buffer before actually being saved to the file being worked on. This works in much the same principle as a bus stop. Only so many people can ride the bus at a time and so those who cannot fit must wait at the stop until another bus arrives.

Byte: A Byte refers to a set of 8 bits. An 8-bit sample requires one byte of memory to store, while a 16-bit sample requires two bytes of memory to store.

C

Card Reader: Used as a medium to transfer files from a memory card to a computer without having to link the hardware the data originated on directly to a computer. Most readers can both read and write to multiple formats, some able to handle from 50 to 100 different card types.

Cardioid Microphone Pattern: The heart-shaped area that defines where a cardioid microphone is designed to pick up sound from. The cardioid pattern itself is what results from tracing the path of a point on one fixed circle as it rotates around another fixed circle. Used mostly in recording vocals and speech, because the cardioid pattern accepts sound far more readily from the front rather than from surrounding inputs.

Channel: Single channel recording is known as monaural or mono recording. Stereo recording involves the recording of 2 channels (left and right). In interviewing situations, the two channels associated with stereo recording allow the separation and isolation of channels for the interviewer and the interviewee. Stereo recording is ideal from a sound quality situation involving two microphones and two entities. Single-point stereo microphones involve the left-right separation but do so from a single source, yielding much less sound isolation. Stereo recording doubles the data footprint of your recording. Some recorders are capable of using a stereo microphone setup but recording in mono. The recorder “mixes” the two channels together. The resulting file is half the file size of a stereo recording, but the channel isolation will be lessened. Some portable recorders will not allow you to record mono files.

Channel Convert: A process that converts audio from either stereo to mono or mono to stereo. Also can be used to transfer bits of stereo audio from the left channel to the right or the right to the left to create fading effects.

Clipboard: Area on a computer where any information copied or cut is stored. Generally anything cut from one area in a program or operating system can then be pasted into another, provided that it remains in the clipboard. For audio data to be pasted from a cut or copy, the program that is receiving the sound file must be able to handle sound data from the clipboard.

Clipping: Occurs when the level of sound in an audio track or file is above the highest recording level available. If clipping occurs then it will cause distortion in audio playback.

Codec: Software solution that compresses and decompresses an audio file for digital playback. The goal of a particular codec is to compress a recording in order to make the file size smaller. Examples of codecs include codecs include Mp3, WMA (Windows Media Audio), Real Audio.

Compact Flash: A flash based mass storage medium that is used in portable recorders. The capacity of this device is ever increasing as are the transfer speeds. Transfer speeds do not affect audio recording, however, they do greatly impact the speed at which the recorded interview can be transferred to the computer. Compact flash is the larger format of the flash based media being used in portable field recorders and involves matching the pins in the device to the holes in the card. Although though it is rare, pins on recording devices can be bent thus rendering the device unable to recognize the CF card and therefore not record. Because of digital photography, this format is quite common in the marketplace and extremely inexpensive in the 2-4 gb range.

Compression

Compressing an audio data file involves compacting the data into a smaller filesize. "Lossy compression" (such as mpeg compression) involves an algorithm that applies psychoacoustic principles to determine what can be taken out of an audio signal in order to make the file smaller. This involves degrading the signal. Although these algorithms are making compression more and more imperceptible, however, if any derivative files are made from your master recording (that was originally recorded using compression) digital artifacts will emerge and the derivative files will be extremely low quality.

Compounding compression–or recompressing an already compressed audio recording, will seriously degrade audio quality. Flash and HDD storage have dramatically dropped in price and therefore, compressed recording is, again, not recommended for interviewing. Most common for audio is the Mp3 format, however, Marantz has used Mp2 on some of their recorders.

MPEG recording will dramatically decrease your data footprint and thus increase your recording time. The consequence of this compression will degrade your recording quality, however it is ideal for web deliverable files. Mp3 files have become a standard uncompressed codec almost universally accepted by most computer players. Mp3 files are usually measured by "bit rate" rather than by sample rate and bit depth. Also included among more proprietary audio compression codecs are the Windows Media Audio Files, and Real Audio file. There are several "lossless" compression codecs available including FLAC (Free Lossless Audio Codec) and the Apple Lossless Audio Codec (ALE/ALAC), however, at this time, most recorders to not include them as a recording option.

Compression Ratio (Data): The ration of the size of the uncompressed file prior to compression to the size of the file following compression. A 14: 1 compression ration means that the file is 14 times smaller than the size of the uncompressed recording, or if the same recording had been originally recorded without compression.

Condenser Microphone: In a condenser microphone a capacitor is used to sense changes in the distance between two metal plates as the vibrations from a sound source hit them. The changes in distance are measured by the minute fluctuation of an almost constant electric current running between the two metal plates. Since the capacitor requires and electric charge, condenser microphones require a power source, either from batteries or phantom power supplied by an audio jack.

D-F

Digital-to-Analog Converter (DAC): These converters are found in any hardware that plays audio from a digital source. The most common being CD players. Since the data stored on the CD itself is digital and the sound that we hear from the speakers are analog, the DAC has to take the digital information provided from the digital source and convert it into an analog one.

Theoretically the process can be completed without any degradation in audio quality from the digital source to the analog one, assuming that the appropriate amount of bandwidth is provided. However, actually achieving this lossless quality is practically impossible due to the quantization error, where the numerical values assigned to the data being transferred do not quite equal those present in the analog audio signal.

Digital Audio Workstation (DAW): A DAW is a system that allows a user to play, record and edit digital audio. Can come both as a computer software program or as an all in one audio workstation, though these have become less popular as personal computers have become more advanced and cost-effective.

DC Offset: This occurs when an outside source causes the mean value of the amplitude of a waveform to be either higher or lower than zero. When normalized a waveform with no DC offset will be able to maximize its volume, but if affected by DC offset, it will have lost a variable amount of headroom for volume.

Decibel (dB): Sound is measured in decibels, which measures audio intensity. Named after Alexander Graham Bell. Digital recorders measure levels up to 0dB. Levels exceeding 0dB result in clipping and distortion. Good recording levels in an oral history interview will range from -12dB to -6dB, leaving enough headroom for intermittent spikes in levels.

Device Driver: The device driver supplies the information needed for a computer to exchange information back and forth between itself and outside hardware. These drivers are almost always provided with the hardware that requires them and manufacturers provide these drivers on their websites free of charge. This system allows for hardware to be updated on a regular basis, providing better compatibility with and better content from said hardware.

Destructive Editing: Refers to editing within a digital audio program where any changes made to the file are processed when executed. For example, if a cut is made from a file in Sony Sound Forge the cut is processed to the file that was loaded at the time. This is the opposite of **Nondestructive Editing**.

Digital Audio: This is the use of digital signals to reproduce sound. This digitization allows for the easy storage of audio files, but also for the manipulation of them since they are not limited by time or amplitude as analog sources are; digital audio breaks down sound from a continuous line of audible information into a segmented, digital representation of said line allowing the user to access and change each bit individually. The higher the quality of a file (the higher the sample rate) the smaller each segment created is and so the closer a file is to the original analog version.

Digital Audio Editor: A digital audio editor is software that allows a user to visualize audio files stored on a computer as well as manipulate them. Files can be changed between formats, volume levels increased or decreased and a myriad of other options are available depending on the software chosen.

Digitization: This is the process of representing an analog source as a discrete set of points (samples). As analog sources are continuous in all respects, digitized versions of them are approximations of the assumed values actually represented in the analog original.

Digital Signal Processing (DSP): This is a blanket term that applies to any alteration of digital data. Analog signal processing has been around for decades in hundreds of applications, but the advent of the digital age has allowed a computer to do far more with signal processing due to the discrete nature of digital data. DSP exists as a method of measuring and changing real world analog data and so must undergo an analog to digital process as well as a digital to analog one at the end of processing. This is a time consuming process, but digital signal processing allows for far greater control as well as safety than does analog.

Digital Signal Processor: A microchip developed to exclusively handle the mathematically intense operations required by digital signal processing.

Directional Microphone: These microphones focus their polar pattern (the area they are intended to pick up sound from) in specific ways. They can be bi-directional, i.e. a stereo microphone, which will record to the side, rather than all around the microphone housing. Others include shotgun microphones which are used to focus on specific sounds, while essentially eliminating unwanted side and background noises.

Distortion: In audio, distortion is a negative term unless you play the electric guitar. It refers to any unwanted change to the original waveform due to, clipping, compression, aliasing, mixing, modulation, power supply problems, as well as changes during the analog to digital conversion process.

Drag and Drop: Allows for easy manipulation of text or other selectable material on a computer. To perform, simply select an area of the information you wish to move and, while the mouse pointer is over the selected area, press and hold the left mouse button. While holding, the selection can be moved to wherever you wish to place it within the current program. To let go of the selection, just let go of the left mouse button.

Dynamic Microphone: Works by harnessing the power of electromagnetic induction. Well suited for onstage environments because they can support levels of high gain before succumbing to feedback, these microphones are popular and relatively inexpensive.

Dynamic Range: The ratio (difference) between the highest and lowest levels of any changeable quantity, including sound.

Electret Condensor Microphone: Work under the same principles as normal condenser microphones, but the electret variety are made with permanently charged material to read sound waves, thus eliminating the need for polarizing voltage. Most of these microphones do require a battery, but the power drain is so minimal that they may have an integrated one or only call for a single AA.

Equalization (EQ): The process of raising or lowering the frequency response attributes of audio through active or passive means.

File Association: The process by which the computer links certain types of files by their extensions (.wav, .mp3, .txt, etc.) with the programs that can open them.

File Format: The type of file saved on a computer. When dealing with audio on a Windows machine, .wav is the most common format.

File Size Limit: A limit on the amount of space that a single file on a computer HDD may occupy. Limited by many factors including program and operating system. The most encountered size limit on a Windows computer is 2GB.

Filename: Name given to a computer file. Usually chose when a file is saved. The file type is usually denoted by a three letter/number suffix on the end, i.e. .mp3.

Firewire: Commercial term referring to an IEEE 1394, a data connection used for high speed transfer of data including digital audio and video. Firewire is the brand name for Apple Inc.'s IEEE 1394. The original incarnation of Firewire is now known as Firewire 400 because of its transfer rate. New innovations now utilize Firewire 800 connections which allow for significantly faster transfer of data than USB 2 interfaces. For Sony products IEEE 1394 is known as i.Link.

Flash Memory: Non-volatile memory used to store data in thousands of applications. Allows for the easy transport of gigabytes of data and contains no moving parts, adding to the safety of flash memory systems.

Flat Response: This is the characteristic of any audio system that can reproduce an original tone without deviating from the original levels of intensity in frequency.

Frequency: The number of occurrences of a repeating event per unit time.

G-H

Gain: The ability of an electronic circuit to increase the power or amplitude of an electronic signal. In Audio this applies directly to the volume.

HDD (Hard Disk Drive): A non-volatile data storage system found in almost all computers. Stores data on rapidly rotating magnetic platters. Now available in Gigabytes and in Terabytes.

Headroom: This is the amount by which the ability of a system to reproduce quality audio (before clipping or distortion) exceeds a certain point.

Hertz (Hz): A measure of frequency per unit time or cycles per second.

High-Pass Filter: Filter that allows high frequencies to pass through, but catches (attenuates) low frequencies at a certain point allowing for the easy elimination of unwanted lower frequency noise.

Hiss: Noise present in many situations that can be reflected in recorded audio. In digital audio the biggest threat is the hiss that all electric circuits exhibit. Can be audible or not, but always has the ability to ruin a recording.

Hold Button: Found on most portable devices, especially audio hardware. Use of the hold button will lock all the other buttons so that they won't be pressed in error. Usually present in slider form, but sometimes as a button itself.

"Hot:" Term generally applied to audio that is very loud or exhibits high, active tendencies when viewed on a level meter.

Hum: Sound produced when audio equipment picks up the oscillation of electronic currents from other equipment nearby. The hum is audibly reproduced through speakers.

Hyper Cardioid Microphone: Much the same as a normal cardioid microphone, but features a highly tightened cardioid pattern. This focuses the sensitivity pattern into a tight area to the front of the microphone.

I-L

Internal Microphone: Microphone that is housed within audio recording equipment. While convenient, these microphones are sometimes incredibly limited in their abilities.

Key Lock: See **Hold Button**.

Kilohertz (kHz): 1,000 Hertz (Hz). Unit of measure for sample rate in digital audio.

Lapel / Lavalier Microphone: Small dynamic or condenser microphone that is usually clipped to a lapel or shirt to provide convenient recording of speech. Because of the proximity to the sound source, these microphones usually produce a much better signal to noise ratio.

Level Meter: Graphic display representing the loudness of sound during recording or playback. Generally color coded and give a warning when sound levels are peaking.

Limiter: A Limiter is a compressor that can be used during recording to make sure that sound levels do not peak. Essentially creates an input volume limit by blocking frequencies over a certain amount.

Line Level: This describes the signal strength of audio passing between two hardware components, i.e. from a CD player to an audio receiver.

Low-Pass Filter: The opposite of a **High-Pass Filter**. Allows signals of low frequency through while attenuating those of higher frequencies set by the user.

M-O

Marker: A user deployable point for quick reference and easy editing in digital audio programs as well as tape decks and CD players.

Microsoft Sound Mapper: Device built into Windows that will automatically select the best sound card or system to play sound on your computer.

Mid-Side Recording: Recording technique utilizing a microphone pointed directly at the speaker (mid) and one at a 90 degree angle (side) to then mix the two audio channels into a traditional stereo image.

Mini Jack: See **TRS (tip, ring, sleeve) Connector**.

Monitor: Act of maintaining watch over sound levels when recording. The dynamic nature of human vocalization makes monitoring incredibly important.

Monitor (speaker): Used in studio applications as speakers. Studio monitors tend to produce a flat response and ensures that the audio heard by the user through the monitors is as close to the actual sound.

"Muddy": Describes poorly defined sound quality where all the frequencies tend to run together.

Noise Floor: Refers to the measure of audio signals present when no direct input is being received by the system. Usually this is noise created by the system itself, by any number of mechanical and electrical systems. Typically, inexpensive recorders usually have a higher noise floor than more expensive, professional level counterparts.

Nondestructive Editing: Opposite of **Destructive Editing**. When edits are made to an audio file in a program that uses nondestructive editing, the audio file in question is not actually changed. The program just places markers in the file telling it to skip certain portions or to make whatever changes have been made.

Normalize: Act of increasing volume to allow the highest level sample in a file reaches the top of a user defined limit. This ensures that the file will utilize the as much dynamic range as possible.

Nyquist Theorem (The Sampling Theorem): During analog-to-digital conversion, the maximum frequency that can be represented by a digital audio source equals half the sample rate used to capture the analog audio. If human hearing is only capable of perceiving 22.5 kilohertz, the sample rate must equal 44, 100 samples per second   or 44.1 kilohertz. This explains why "cd quality" was originally set with a sampling rate of 44.1 khz.

P-Q

Pan: The act of perceptually moving a mono sound signal across stereo or other multi-channel speakers.

Pause: Control that stops audio playback but allows for continuation at the exact point the pause occurred. Important in digital recording as pressing stop while recording will usually create a whole new file, but pause will only stop the recording at a certain point.

Peak Levels: See **Permitted Maximum Level (PML)**.

Permitted Maximum Level (PML): Highest volume level in a given program of piece of hardware. Any audio input that surpasses the peak will result in distortion or “clipping.”.

Phantom Power: Used to send electrical power from the recorder through the microphone cable to a condenser microphone–which requires Phantom Power to operate. True phantom power is signified by 48 volts and is provided by the recorder when enabled.

Polar Pattern (microphones): Refers to the sensitivity of a microphone in a given direction. Common patterns include Cardiod, Omni-Directional and Shotgun patterns.

Proprietary Format: Proprietary format is usually a digital file technology that is owned by a particular corporate entity. Working in proprietary formats at either the recording or the archiving phases is strongly discouraged as these formats discourage interoperability and may not be accessible of the corporate entity abandons the technology.

Preamplifier (mic pre): A circuit on recorders with microphone inputs that boosts the analog audio signal from the microphone enabling the recorder to record the audio signal. Lower quality microphone preamps will add noise as it boosts the signal and are found on many lower end recorders. Higher quality microphone preamps add far less noise leading to a more “transparent” signal boost. Preamplification on a portable field recorder takes place prior to analog-to-digital conversion.

Pulse Code Modulation: Most portable field recorders and a/d converters utilize a representation called Pulse Code Modulation (PCM) for digital conversion of an uncompressed analog signal. This digital data is then saved as a data file (either .wav or .aif).

Quantization: The process of assigning approximate integer values to analog measurements. This process occurs every time that an analog file of any type is turned into a digital one. As far as audio recording goes, the higher the quality of the digital recording, the greater the number of integers used in the approximation, which results in less **Quantization Noise**.

Quantization Noise: This is the hiss-like noise that results from Quantization during an analog to digital conversion. A result of all digital recordings because they are based off of approximations of the original analog values, not exact copies. The higher the quality of a recording, the less noise present in playback.

Quarter Inch Input: A common **TRS** connector commonly used for applications ranging from headphones to musical instruments to microphones.

R-S

RCA Connector: Used in many audio and video applications to transmit analog signals. Name comes from the RCA corporation who introduced the original design.

Record Level: The record level refers to the current loudness of an incoming audio signal. Usually, physically represented on a level meter.

Record Indicator: Used on almost all recorders to alert the user that recording is indeed taking place. In newer recorders there tend to be two or more indicators used at a time. Sometimes a red light will come on as well as an on screen symbol or word to represent recording.

Record Pause: Pressing the pause button during recording on most players will allow the user to resume recording right where he or she left off, rather than having to stop and then create another file.

Resample: A process by which a sound file is sampled at a different rate than what it was originally recorded. When dealing with resampling it is always best to make your original recordings at the highest possible bit and sample rate and then to downsize these files later on if need be. While moving from a 16-bit file to a 24-bit file will not result in a horrific loss of clarity, you will still be left with a sub-par 24-bit audio file.

Sample Rate: Refers to the number of samples per second in a given digital audio file. The higher the sample rate the greater the sound quality, but the larger the size of the file.

Secure Digital (SD) Card: SD cards are one of the most common flash based memory cards on the market. They range from 4MB to 4GB on the SD standard and from 4GB to 32GB on the SDHC standard. Always make sure that your recorder or card reader is compatible with the standard you have chosen.

Shotgun Microphone: A highly directional microphone, focusing almost the entirety of its **Polar Pattern** directly in front of the microphone.

Signal-to-Noise Ratio (SNR): Refers to the ratio between a recorded signal and the noise levels associated with it. The higher the signal-to-noise ratio the better the quality of the audio.

Sound Card: Sound cards are found in computers and allow audio from the computer to be heard. They have become less and less popular in digital recording because their proximity to the computer equipment (being inside the case) causes a lot of electronic noise to present itself in audio files. It is far more common and a better practice to use an outside audio interface.

S/PDIF: A standard developed by Sony and Phillips to move digital audio signals from one place to another. Generally uses coaxial or fiber optic cable

Stereo: Refers to audio in which the sound has been separated into two or more distinct channels.

Stereo Microphone: Actually two microphones in one housing, this configuration is well suited for interview recording as it will separate the interviewer and interviewee between audio channels when placed correctly between them.

T-V

Threshold: Refers to a level where a signal processor will begin to act on a signal. Thresholds are set at different levels depending on the application, i.e. a high-pass filter would attenuate sounds below a certain threshold.

Transducer: the conversion of acoustical energy (sound waves) into electrical energy. A microphone is a transducer

Transparency: Typically associated with higher end preamps which boost audio signals without adding significant noise.

TRS (Tip Ring Sleeve) Connector: These are the typical connections for audio equipment, including microphones and headphones. They come in three different sizes: 3/32″, 1/8″ and 1/4″.

Uncompressed Audio: This refers to audio that has had no compression applied to it and is therefore as close to the original sound as possible, though still affected by the sample and bit rate of the recording. The most common format is the .wav.

Universal Serial Bus (USB): A connection standard for connecting hardware to a computer. Allows for hot-swappable connectivity, where an item can be plugged in or unplugged without turning the host computer on and off again. Originally transferred data at 12Mbit/s, but now transfers data at a rate of 480Mbit/s with the 2.0 High-Speed standard. Theoretically very close to the speed of a Firewire 400 cable, USB 2.0 is still noticeably slower in intensive data applications such as external **HDDs**.

Volume: Standard term for the loudness of an audible sound.

W-Z

"Warmth": A term that describes the sounds resulting from the 150-400Hz range.

.wav: File standard developed by Microsoft and IBM. Most common of the uncompressed audio file formats.

XLR: XLR inputs are the highest quality analog inputs. The connection is a "balanced" signal. A balanced connection enables the linking of analog audio devices, including mics, to a recorder through impedance-balanced cables. Usually associated with professional level audio equipment, these allow for longer cable lengths and reduce the addition of external noise to the signal.

Balanced cables have either XLR or TRS plugs. Professional level digitization will usually involve balanced outputs on the analog playback device and balanced inputs on the analog to digital converter. XLR cables transport a mono signal. Stereo recorders with XLR inputs will contain 2 XLR inputs. Single Point Stereo microphones will contain a modified version of XLR, in certain cases, a 5-pin connector that splits into two XLR male cables.