Glossary for the YAMAHA AW Audio Workstations

A/D converter:
The A/D converter is responsible for the conversion of analog signals into digital signals. The analog signal is sampled every few milliseconds and its level is quantized into a digital word… the larger the digital word, the more accurate the representation of the analog value. The AW1600/AW2400 use 24-bit linear analog-to-digital converters for its analog inputs

Absolute time:
Refers to the start point of a recording. It is the time code stamp (and cannot be changed).
See also Relative time.

Automix: (amx) (AW2400 only)
An Automix is one of sixteen automated mixes that can be stored with each SONG. It represents your fader movements, mute on/off, EQ settings, pan positions, and Scene changes. You build up your Automix via multiple passes. You experiment, figure what you want to do, then enable the AUTOMIX REC and document your changes. This allows you to do everything without the pressure of having to do it all in a single pass. That is the key advantage of automation - you can accomplish complex changes at your leisure. There are two types of record: Automix [AUTO REC] and [REC] - in [Auto Rec] mode you can use the [SEL] buttons to activate the channel's automation at anytime. In [Rec] you select the track to update ahead of time.

- [AUTO REC] does not need to be re-armed each time and therefore is best when you are automating a series of mixing tasks. The unit remains in automated record when transport is stopped so you can resume immediately.
- [REC] will end when the main transport is stopped.

See also SCENE for alternate automation.

Auto Punch:
See Punch In / Punch Out

Backup:
Your Song data can be stored in digital form (not audio) to either internal CD-R, computer or external USB devices via the computer (it does not write directly to USB drives but must backup to the computer). This is for long-term storage and for safety purposes. A Backup can be restored. It is the entire multi-track recording, including the virtual tracks.
See Restore

Bounce:
The word is used as in 'bouncing tracks'. This is when tracks are combined in a submix. With the AW you have a flexible system that allows you to take signal returning from the HDR and route it through a channel, applying new EQ and/or effects, then re-route it, via an available bus, to another track(s) on the HDR. In this fashion it is possible to combine multiple tracks to a stereo pair, for example. Typically, this could be used to combine 6 or 8 drum tracks down to a stereo pair.
Bounce is used during the Pitch Fix (pitch correction) function. The Source track is processed and sent via the Bounce function to a Destination Track or Virtual Track – an Auto Punch in and out is used to define the start and end of the pitch correction processing.

Bus:
First, it is 'bus', b-u-s. This is a case where the word is a very good analogy. A bus carries something(s) from here-to-there. In this case, it is audio signals. And like a bus, it is capable of carrying multiple signals from one place to another. We refer to, "routing signal from channel 1 on bus 8 to track 16".

Channel:
Channel refers, specifically to a physical 'input' row or 'track' row of controls. You have input channels and track channels. For example, if you have an AW2400 you will have 12 channels numbered 1 through 12 and via a fader flip, channels 13-24 share the hardware of channels 1-12. Input channels are pre (before) the recorder. You will also have 24 identical track channels through which audio is played back post (after) the recorder. The significance of the difference is as follows: During the RECORD process changes made to the EQ, Dynamics processing or levels on the Input channel will be recorded to the hard disk, while changes made to the EQ, Dynamic processing or level on the Track channel will be monitored only. During a MIXDOWN both the Input Channels and the Track Channels can be sending to the final stereo mix. Whatever changes you make will be documented. In the AW1600 you have 8 Input Channels and 16 Track Channels plus 4 sampling Pad Track Channels.

Compander:
See Dynamic Processor

Compressor:
See Dynamic Processor

Compressor Library: (AW2400 only)
The AW2400 has a separate Compressor per Channel. There are 36 preset programs (001 to 036) for you to recall and 92 programs (037 to 128) for you to store your own Compressor settings. Library data can be recalled from any session (Song) on your HDR.

Cue:
CUE or TRACK CUE lets you hear (monitor) the signal directly from the recorder. This bypasses any panning, effects, EQ, etc. that you have added to the monitor mix.

D/A converter:
The conversion of a digital data stream into analog signals; the digital word is buffered and then converted into an analog signal. After conversion, the analog signal is usually processed through a smoothing filter, which removes the step transitions between the digital words. The AW1600/AW2400 use 24-bit linear digital-to-analog converters for its analog outputs.

DAT cassette:

Metal chrome cassette tape (thickness 13 (m, width 8.8mm) replaced by the STEREO TRACK in the AW environment. You will no longer need to mix to DAT before mastering your CD.

Display:

Display refers to the numbers in the time area of the display. It can read seconds, Time Code or Measures. This can be set in [SONG], [F2] Settings. Seconds is regular clock time. Time Code is MIDI Time Code: hours, minutes, seconds, and frames. And Measures is for musicians - if you intend to use measures you should either record with the provided metronome as a reference or use an external MIDI sequencer under sync control of the AW - this way your measure lines are accurate.

Dither:

Dithering is a mathematical process where a random noise is added to the least significant bit of a digital word. With very low-level signals, the quantization error becomes correlated to the signal level. This creates a measurable amount of distortion. By adding dither, the correlation between the signal level and the quantization error is canceled, allowing the digital system to encode amplitudes smaller than the least significant bit. If you change the word size as a signal passes from one digital system to another, being able to add dither allows you to maintain a high quality signal.

DSP (Digital Signal Processor):

A specialized circuit, usually a chip, that is designed to manipulate large quantities of data in real-time. YAMAHA has developed world-renowned expertise in proprietary musical application specific digital signal processing.

Ducker:

See Dynamic Processor

Dynamic Processors:

These are configurable devices that will help the engineer control the level of incoming signal (control dynamics). The units (one per input/output channel) can be configured as a compressor/limiter, ducker, expander/gate, or compander. Basically, to explain these best: they are actually amplifiers. Many people are surprised by this description but they are specialized amps that do not just get louder. The AW2400 adds a dedicate GATE for each input channel – this can be put to great use when doing live recording of drum microphones to reduce leakage from microphone inputs.

- **Compressor** - A **Compressor** is an amplifier that when signal exceeds a programmed level (called the 'threshold') the device then reduces the gain by a predetermined 'ratio'. In plain English: If you set the threshold at -4dB and the ratio to 4:1 compression, then for each 1dB the signal wants to exceed -4dB it will be reduced by a 4 to 1 ratio. Therefore, if enough signal were to hit the threshold of the compressor to normally send the meter to 0dB the compressor will reduce the dynamic range and only let the meter reach .3dB (or 1dB for every 4dB it want to move). A compressor has the effect of reducing the overall dynamic range because the sound will not get louder at the same slope, beyond the threshold. It will sound as though the softer sounds - those below the threshold point (which are unaffected), will now sound closer in level to the loud effected sounds (those above the threshold). On a subjective level the sound is said to be 'fatter'. The difference between a compressor and a limiter is simply a matter of degree.
- **Limiter** - A **Limiter** (compression ratios of 10:1, 20:1 and beyond) is an amplifier that puts a ceiling on how loud a signal will get. Once signal reaches the threshold level it is 'limited' and will not exceed that level. Using our -4dB example above, if the ratio was 20:1 then it would take a signal that would normally reach +16dB (absurd) in order for the signal to now reach -3dB (20dB above threshold for and increase of 1dB) and +80dB (beyond absurd) to reach 0dB. When a signal is over-compressed or limited it is said to be 'breathing' or 'pumping'. This saying comes from the phenomenon where the intake of breath by a vocalist (normally a soft sound) is now so close in level to the normal speech or singing it sounds as if the breathing is bigger than life.
- **Expander** - An **Expander** is an amplifier where the signal is suppressed until the signal reaches the threshold. This is said to expand the dynamic range by lowering the noise floor. If background noise is a problem an expander's threshold can be set so that the signal below it are reduced. The difference between an expander and a gate is one of degree, as well.
- **Gate** - The **Gate** or Noise Gate is an expander that ignores signal below the threshold. This, typically, is where an external signal can be used to initiate the action, called 'keying'.

- **Ducker** - The **Ducker** reduces the level of one signal when another signal is present. Typically, it is used to lower the music when a speaker is talking.
- **Compander** - The **Compander** combines the compressor and the expander. It has a parameter called 'width' that lets you set the distance, in dB, between the compressor and expander. Very soft sounds are suppressed and very loud sounds are reduced - the width is the dynamic range you will use.

With any of these specialized amplifiers you can use the 'keying' feature to trigger the response. In addition, these devices all have time parameters that control how quickly they engage (attack) and disengage (release) the signal. You also get an output gain control (they are amplifiers) to return any lost level or reduce any excess. Remember you are controlling dynamic range - the distance in dB between LOUD and soft. Once you have it set you can then raise or lower the entire 'range' as a unit. This is how to let a soft sound (like a voice) be heard over an entire big band. Favorite settings can be stored and independently recalled from Songs.
See Dynamics Library.

Dynamic Range:

The difference between the loudest (maximum output level) and quietest (residual noise floor) sounds produced in an audio system. The dynamic range in a digital system is determined by the data resolution, about 6 dB per digital bit. A 24-bit system has a theoretical dynamic range of 144 dB. The AW has a specified dynamic range of 109 dB.

Dynamics Library:

An area in AW memory used to access and store dynamics settings - stored as programs. There are typically 40 preset programs (1 to 40) for you to recall and 88 or more user programs (41 to 128) for you to store your own dynamics settings. Library data can be recalled from any session (Song) on your HDR.

Edit:

Edit features of the AW are fairly extensive. You define your area to edit as follows: Track, Region and Part. A track is an entire single recording area from beginning to end. A region is any continuous recording, in-point to out-point. A part is any area that you define, continuous or non-continuous. Tracks can be named, copied, erased, exchanged, slipped (move forward or back in time); time-compressed and/or pitch shifted (without tempo change). Regions can be user defined by using the DIVIDE function and they can be named, copied, erased (leaving a hole), deleted (closing the gap), and/or actually moved to other locations. The Region TRIM IN and TRIM OUT functions allow you to make minor changes to start and end points (principally to remove clicks or ticks). Data imported from CD or via USB can be imported into a Region. A Part can be set up for convenience. It differs from a Region in that it can incorporate more than one region within a track. In addition to erase, delete, copy and move, parts can be inserted moving other data back by its length. An area defined as a Part can be time compressed and pitch changed. TRACK EDIT and SAMPLE PAD EDIT are the two main areas for editing on the AW.

Edit Buffer:

The Edit Buffer is the current settings. When a Scene memory is recalled, for example, the mix settings of the selected Scene memory are written to the current Edit Buffer. When a Scene memory is stored, the mix settings in the Edit Buffer are written to the selected Scene memory. Your most recent activities are stored in the Edit Buffer (15 deep) - letting you recall most of your recent activities. You must clear the buffer with the Optimize function before mastering a Song.
See Optimize.

Equalizer:

The AW is equipped with a four-band, fully parametric equalizer, with variable bandwidth (Q), center frequency (F), gain (G), and ON/OFF parameters. Equalization can be applied to the input channels (for recording), the HDR tracks (for playback), effect return channels, and the stereo output. For stereo channels (paired), the equalization curve is applied equally to both channels. See also TRANSPARENT; EQ TYPE I and II

Equalizer Library:

An area in AW memory used to access and store equalizer settings - stored as programs. There are typically 40 preset programs (1 to 40) for you to recall and 88, or more, user programs (41 to 128) for you to store your own equalizer settings. Library data can be recalled from any session (Song) on your HDR.

EQ Type I and II (AW2400)
EQ Type selection found on the AW2400 allows you to select between an accurate digital EQ (Type I) and one modeled after an analog style EQ (Type II). Both are four band parametric types, however, Type II has the warmth and character not heard since analog Equalizers were popular.

Expander:
See Dynamic Processor

Fader:
A fader is like a valve on a pipe. It, when open, allows signal to flow. You can close it to stop the flow of signal. If a send is pre-fader it will continue in that direction but all things post-fader are now closed off. The Fader is not an input level device (that is called a Gain Trim), although it is often mistaken for it. If the signal is distorted at the Gain Trim, then turning it down at the fader will have no effect on reducing the cause of distortion - you only hear it less well.

Gain/Trim:
Gain/Trim is a combination of terms. Gain is an increase in level and Trim is a decrease in level. And that really says it all. It is responsible for the scientific matching of the signal to the input stage. Mic level are typically -50, guitar (-35), synth (-10) or audio line (+4). In general, increase the gain pot until you see the overload LED light. This will light at .6dB below clipping. You should see it light on the loudest activity. It is okay for it to flash - it is bad for it to stay steady

Gate:
See Dynamic Processor

Gate Library: (AW2400 only)
The AW2400 has a separate Gate per Input Channel. There are 4 preset programs (001 to 004) for you to recall and 124 programs (005 to 128) for you to store your own Gate settings. Library data can be recalled from any session (Song) on your HDR.

Grouping:
There are 2 types of groups you can create: Mute groups and Fader groups. You can have four of each type. When you create a group, any one of those in the group can act as the master for the others. If, for example, you create a mute group with 9 channels in it, muting one will turn all of them off. If you have five drum channels in a particular group, then any one fader will raise or lower them all, in proportion to each other.

Import:
You can ‘import’ or bring in audio data from external sources, such as regular audio CDs (there is a copyright warning) or from .wav files (via the CD or SCSI). Imported data goes either to the sample pads or directly to audio tracks. You can also ‘import’ data from a track to a sample pad - to a maximum of 90 seconds total. When you ‘import’ audio data to a track, there is virtually no limit to the total time (up to the total Remain time on the HD).

Initialize Hard Drive:
If you initialize the AW hard disk from your computer it will become unreadable by the AW. Always use the UTILITY screen “D.IN • HDD” page to initialize the AW hard disk.

Libraries:
There are several areas of the AW for the user to store and/or recall particular setups. When you have used any of the AW features you can store your work in an entity called a ‘Library’. This makes things like the digital Patch Bay easy to use. And more importantly, they are recallable. Libraries for storing EQ, dynamics processor, effect processor and channel settings are provided. As you develop your style you can store your personal favorite settings and they can be recalled from session to session. Library data can be recalled from any session (Song) on your HDR. [SONG] > [F3] Edit. Use Mixer Imp (mixer import) to define the data you need to recall.

Limiter:
See Dynamic Processor

Loading:
Loading a Song is the process of bringing the associated data of a Song from the HD to the current Edit Buffers. Loading is a very short process, approximately 5 seconds.
See Restore.

Markers:
You have markers to point out up to 99 locations in the song, 1-99. Basically, you use a few at a time to designate in and out spots, start and stop points, etc. The 'A' and 'B' markers are particularly useful to designate where you wish to start and end an edit. They will instantly memorize the clock display setting and let you 'paste' the value into the edit screens with a single button touch, as necessary. 'A' and 'B' are different than the others as they are intended to be used temporarily when you are involved in working on a particular section. You will also see markers: 'S' for Start of record, 'E' for end of record, 'I' for punch In point, 'O' for punch Out point, 'R' for relative zero point

Mastering:
The process of making a CD from a SONG session is called 'mastering'. By definition a CD is 16-bit, 44.1kHz. The Mastering process in the AW typically will have you transferring one of your mixes from the Stereo Track to a CD. Once the Song has been OPTIMIZED it can be made into a CD.
See Wav Editing with a Computer

Metering:
The AW features comprehensive signal level metering. The input channels, the HDR, effect returns, and the auxiliary sends and buses are all metered using the METER display function. Control Room Monitor signal is metered, as well. Peak hold is available for all meters.

Metronome:
In conjunction with the TEMPO MAP feature, the metronome can be used to outline complex time signature changes within a song. Having a metronome on a session makes creating a count-off a breeze and comes in very handy when a session will involve overdubbing.

Monitor:
The word has several meanings in the studio. It means to watch or listen, as in the *monitor mix*. This is the mix that during a multi-track record is a rough balance of the recorded data. Since multi track recording is a number of tracks, each optimized as to record level; the monitor mix exists so that we can listen to it with some kind of realistic balancing. Monitor is also used (in the plural) when referring to speakers. As in, 'NS10Ms are flat studio monitors that have become an industry standard.'

Mute:
The activity of the channel [ON] buttons can be recorded in the Automix (AW2400). You can create 4 mute groups that can be activated when necessary. Mute Events can be stored in SCENE snapshots (AW1600).

Optimize:
This AW process clears the edit buffers (Undo) and discards unnecessary data. Before a Song can be mastered (to CD) it must be optimized. [SONG]> [F3] Edit.

Over-sampling:
The input analog signal is sampled at a much higher rate than the normal sampling rate. Using the high sample rate, the digital data may be processed with a very steep slope digital filter. The filter is in the digital domain, therefore unpleasant side-effects, such as phase effects, are eliminated.
The AW1600 uses 64-times over-sampling its analog inputs and 128-times over-sampling its analog outputs.
The AW2400 uses 128-times over-sampling its analog inputs and 128-times over-sampling its analog outputs.

Patch:
The AW has a sophisticated Digital Patch Bay for connecting and busing signal throughout the mixer. The word comes form the old analog 'patch cord' that was used to link tracks and channels to each other and external effect processing, etc. It didn't feel like a session until you had plenty of spaghetti in the patch bay!

Patch Library:
Twenty of your more complex setups can be named and stored in memory as part of a SONG. …Meaning that no matter how complex your setup for the session it is instantly recallable. There is no equivalent in the real world when you use an analog style patch bay you need a piece of paper and a lot of time to accomplish what this feature does instantly.

Peaking:
An equalizer circuit that is used to cut and boost a signal, centered about a specific frequency. Using the bandwidth (Q) parameter, you can widen or narrow the effect of the circuit.

Phase:
Phase is the frequency coherence of a signal. If two signals are out of phase, the trough of the first waveform corresponds with the peak of the second, resulting in cancellation. The AW can invert the phase of the input signals, which allows you to compensate for incorrectly wired conductors and so forth.

Pitch Fix:
Built into the AW1600 and AW2400 is an innovative pitch correction function. Basically, you define a region to which you want to apply the Pitch Fix using AUTO PUNCH IN/OUT, you select a SOURCE and DESTINATION Track and BOUNCE record the corrected data. You can select notes that will sound or you can plug in a MIDI keyboard (CTRL mode) and play the correct pitches in real time. Pitch can be changed directly by interval increments or you can create a chorusing effect (CTRL = NOTE). You can fix the formant (that portion of the vocal that normal does not change pitch – this eliminates "munchkin-ization". Or you can move the formant changing the gender/timbre of the vocal. The whole Pitch Fix routine is UNDO capable. See BOUNCE; PUNCH IN/OUT

PRE and POST:
PRE and POST are important concepts in signal flow. Here are some examples: Whether an effect send is 'pre' or 'post' the fader, for example, will change how it responds if the fader level is changed. Whether an Aux send that is being used for the musician's headphone mix, is 'pre' or 'post' the fader will determine if your levels in the control room are different from theirs. When you are setup to record, your Input channels, 1-24, are PRE the recorder (they represent what is recorded to multi-track), and your Monitor mix is POST the recorder (you can experiment with effects, panning, etc. because it is only in the monitors).

Program Change:
A MIDI message that is used to recall programs… In the AW they recall Scene memories.

Punch In / Punch Out:
You can automate record start and end points to any location. Using the Markers to define the points you can have clean, accurate punches every time. Freeing you to play your instrument or concentrate on the music, rather than the mechanics of the punch in spot or worrying about writing over good stuff at the out point. You can even automate the pre-roll and post-roll times. You can 'rehearse' your punch in and punch out, ahead of time by simply activating the AUTO-PUNCH feature and hitting PLAY instead of PLAY+RECORD. This is used in conjunction with BOUNCE and PITCH FIX to accomplish pitch correction.

Q (bandwidth):
The bandwidth of an equalizer band - For high values the bandwidth is narrow (10.0). For low values, it is wide (0.1).
See Equalizer.

Quantization:
Used in digital recording when referring to the word length, 16-bit, 24-bit and 32-bit. 24-bit sessions are automatically converted to 16-bit when you burn a CD. 32-bit wav files are stored on the hard drive (all effect processing in the AW1600/2400 is done at 32-bit) it can be converted to 24-bit for exporting and dithered to 16-bit when mixing down.
For 24-bit songs, the audio data contained in the "Audio" folders is stored in 32-bit WAV format. If your waveform editor application cannot handle 32-bit data, first export the WAV file(s) to the "Transport" folder. WAV files imported or exported via the "Transport" folder are automatically converted to 24-bit format that can be edited using most waveform editor software.

Relative time:
Can be set by the user so that if the song begins some time after recording you can set the RTZ or Return to Zero point equal to the RELATIVE time. Absolute time cannot be changed but Relative time lets you set the 0 point. An 'R' marker will appear on time line when you have set a relative time reference for zero.

Restore:
Restoring a Song is the process of moving all the data for a Song from CDROM to the internal Hard Drive. It is a lengthy process because it includes all the multi-track data, the edit buffer data, the Automixes (AW2400), the libraries, etc.
See Loading; Backup

Sample Rate:
AW can handle either 44.1kHz or 48kHz sample rates. If your intention is to burn an audio CD your session should be set to 44.1kHz sample rate. (All consumer CDs, by definition, are 44.1kHz). All data within a song should be the same sample rate.

Sampling Pads: (AW1600 only)
The Sampling Pads offer the user a way to 'fly-in' audio data. Data, up to a total of 47 seconds at 16-bit and 29 seconds at 24-bit can be placed on the pads in one of the following ways:
- Transfer data from a region of a track
- Import data from an audio CD
- Import a .wav file from CDROM or USB

There are four banks of 4 pads. The chief function of the pads is that you can trigger them and the timing data of the triggers can be recorded and become part of the mix. Or you can trigger sounds in a live situation on cue. This lets the engineer become part of the performance. There is some limited editing on board, but principally you will prepare your sample elsewhere (like in WaveLab, or TWE) and then import it to the pads. From there you can use it live or record the performance to a track. Loop phrases can be created, then saved as .wav files and imported into the AW you can then use the 'cut and paste' method to recreate the loops, as necessary, outlining the backbone of a song on the linear track. You can even insert intervals between repeats. In another scenario you could use the Pads to store a number of sound effects. Then as the song develops, you add several 'overdubs', you determine that you want to intuitively enter these sound effects on a track of the multi-track. While the other tracks play, you can 'fly-in' your effects and actually record the results to at track (or keep the 'performance' as trigger events that are mixed directly to the stereo mix). If you print the performance to a track, of course, you can then assign new samples to the pads and continue. The pads are a creative tool that you have at your disposal.

Scene:
A Scene is a snapshot. The concept is like a photograph of all the settings. It is particularly important to create a starting condition for your Song. In other words, what tracks are active, what levels are they at, what effects are active at the very beginning of the mix and so forth are all memorized in a Scene. Digital settings are memorized (that's all the mixer settings with the exception of the analog gain stage). You can recall complex setups and go from one to the other, instantaneously. Scene changes are documented in the Automix and should be acoustically silent. You can look at and edit a list of when Scene change events occur. You have 96 Scenes in the AW1600 (99 Scenes in the AW2400) per Song and they can be recalled at anytime manually or via the automation. Each can be named for easy identification. A semi-automated mix can be created in the AW1600 by placing SCENE recall events on the TEMPO MAP.

Setup:
The AW Setup function has to do with Patching In and Out of the digital mixer (both analog and digital) and setting up the Solo function. You can determine what input arrives on which channel of the mixer and how the optional I/O boards are handled by the system.

Shelving:
An equalizer circuit that is used to cut and boost a signal above or below a specified frequency high and low band equalizers are usually shelving type. The AW equalizer can be configured as shelving or peaking.

Signal to Noise ratio (S/N):
This ratio is the difference between the nominal signal level and the residual noise floor usually expressed in decibels (dB). That is, the difference between normal record level (signal you want) as set per instrument and any hiss in the system (signal you do not want)…expressed as a ratio, dB.

Slice:
Slice is a Quick Loop Sampler function that divides a sample (whose playback mode is set to Loop) into eight to sixteen segments, and adjusts the timing of each segment so that it can be played at a different tempo without affecting the pitch.

Song:
Each AW project is called a Song. A Song contains your multi-track recording, up to 16 Automixes (AW2400), 96 Scene memories (AW1600), 99 Scene memories (AW2400), a Stereo Track (that is, the final mix – times 8 virtual), and up to 16 samples stored on the sampling pads. All your libraries are stored, as well (EQ, Effects, Dynamics Processors, Input, Sample, etc.). Songs can be backed up, restored, and mastered. You can recall entire Songs or certain portions of a Song, for example, you can import a single track from a Song and/or you can recall the mixer settings or libraries from any Song on your Hard Disk.

Sound Clip:
The Sound Clip function lets you record and playback an input signal without affecting the recorder tracks. You can use this as a sketch for your ideas for a song or arrangement. Sound Clip can be used in conjunction with the Metronome and data can then be cut and pasted into a song project.

Start Point
The Start Point refers to the display clock counter. Typically any time the transport is rolling it is referenced to a clock time (displayed as Hours:Minutes:Seconds:Milliseconds or as Hours:Minutes:Seconds:Frames and as Measures and Beats). When you are using an external MIDI sequencer you will want to delay the Start Time so that the displayed time begins at the exact point the music count begins. For example, often in music a count-in is required prior to the downbeat. The START POINT can be moved so that you can create a count-in. By changing the Start Point parameter before a session you can create a 2-measure or 4-measure pre-record area referenced to the Time Signature and Tempo defined on the TEMPO MAP.

STEREO Track:
This is a special track that can be stored with each SONG. You must create this stereo Track in order to complete the project and burn a CD. The best way to think of it is in comparison to a component system. The Stereo Track step is what the DAT tape used to be - it is the final mixdown. It is your favorite mix. The Stereo Track is the one that will be burned to CD. When you activate the Stereo Track, the multi-track inputs are automatically muted. The Stereo Track will playback through channels 1 and 2. Conversely, when you are listening to the multi-tracks, the Stereo Track is automatically muted. There are actually 8 STEREO mixes that can be stored with a SONG project…allowing you to compare different mixes or allowing each member of the band to create a mix version. You select one to burn to CD.

Track:
A Track is a physical location on the Hard Disk that contains your audio data.

Track Cue:
See CUE.

Transparent
A word used to describe an audio component that does not color the sound. Early on in recording this was a theoretical concept (as most analog components did color the sound) but now with digital components is more of a practical reality. Some analog gear was guilty of degrading the sound, while other devices actually did something unique to the sound that became fashionable. For example, an equalizer is a device that is responsible for boosting or cutting the loudness at a selected frequency. Analog equalizers often were guilty of horrific phase shifts and a slight distortion between frequency bands. While the phase shift always remained an unfortunate side effect, the slight distortion that occurred between frequency bands actually became a preferred sound (warm, European-style EQ warmth, etc.). Because component modeling is now a reality you will find items that can mimic to a highly accurate degree those things that endeared us to old style analog gear. EQ Type II (AW2400 only) is one such device. While EQ TYPE I (found in both AW1600/2400) is transparent (does not color the sound), EQ TYPE II (AW2400 only) mimics the old style analog EQ. As another example, the mic preamps of the mixers are as close to transparent as can be possible – this allows the natural characteristics of the musical instrument being recorded to come through.

Transport Folder
This folder is the go-between folder. You can export a track or tracks to the Transport Folder where your USB connected computer can PLAY and/or EDIT the AW data. The individual recording on a track are made into STEMS and can be opened in your DAW software from the AW drive.
See USB (Computer connection)

UNDO History
The AW1600/2400 have 15 levels of Undo

USB (Computer connection)
The AW1600/2400 can be directly connected to a computer via the built-in USB interface. This allows WAV-format audio files to be copied between the AW and computer for convenient management and processing in computer-based applications, and "backup" song files can be stored on the computer's memory media.
See Transport Folder

Virtual Track
A Virtual Track is a physical location on the Hard Disk that lets you store an alternate take. When recording vocals you can record up to 8 virtual tracks and them 'comp' or compile a complete performance by taking the best of each pass.

Wav Editing with computer:
To directly access audio track or pad data the AW must be connected to the computer via a USB cable, after which the WAV files can be directly accessed inside the appropriate folders in the AW's hard drive. You can use a waveform editor application running on your computer to directly edit the AW audio data.

Wordclock:
Wordclock is a sync-pulse, which allows devices to determine where the start of each digital word is. When multiple digital devices are connected together, it is vital that each device knows where a digital word starts and stops. Otherwise dropout or distortion may result. Although most digital interconnection protocols are self-clocking, it is more reliable to use a dedicated line for your wordclock signal. This is especially important in a multi-track environment where up to eight channels of digital data may be multiplexed on one cable.