The "art" of EQ

by Aaron Trumm

EQ can be used in a variety of situations, from live sound to recording to tape to mixing down. Mainly, it should be used to enhance signals that have some problem. The golden rule of EQ is less is more. If something seems fine without it, I avoid EQing it at all. Then, if I do use it, I try to remain subtle. My personal golden rule is nearly never EQ signals going to tape (as in a multitracking situation). I always try to get the original sound on tape, then I can mess with it later. Putting EQ (or any other effect) on tape usually just leads to trouble. The other rule (the silver rule :)) is cutting is almost always better than boosting, especially when fixing problems. For example if a guitar sounds too thin, first try cutting high frequencies and boosting the gain a bit, instead of boosting the lows. The more clutter you can remove from a mix, the better. A better example is I very often cut a bit of high away from hats. Another example is, many times you may not hear something well in a mix...You might try cutting some frequencies in a different track that seems to be interfering, rather than boosting in the track you want to bring out. With these basic rules in mind, I'll tell you my rules when I enter a mixdown session:

3. Rule Of Opposites: Usually, tracks with high sounds, (a high guitar, hats) need cutting in high frequencies and boosting in lower, and vice-versa. This is really only a starting guide, not a rule. Also, sounds that interfere with eachother can be separated in a mix by EQing them in opposite directions.
4. Bass usually needs a boost in the mid range somewhere and sometimes the high. This way it can cut through and be heard on smaller speakers.
5. Kick drums usually need that same mid and/or high boost on a subtle level so they too can cut through on smaller speakers. For hip-hop, kick needs a low end boost, but NOT TOO MUCH.
6. Snare drums always sound warmer with a boost in the low-mid range and some cut of the highs. An annoying CRACK can be softened with this high cut. Sometimes I boost the lows in snares to make them even fatter. But it really depends on the snare sound. The rule of opposites usually applies here. Snare sounds that were thin to begin with I usually warm up a bit, and heafty snare sounds I might thin out a bit.
7. Hats almost never need any EQ if they're recorded clean. Usually an EQing for my hat tracks is to cut highs to get rid of an annoying hiss.
8. Guitars are simaler to snares for me. A thin original guitar might need boosting in mids and lows (depending on what the desired sound is, and what else is present in the mix) or a heafty guitar might need to be thinned out a little by cutting lows and low-mids.
9. Vocals usually like to have a boost in the mids or high-mids, but it depends on the voice. Vocals nearly always get lost amongst guitars...a good way to deal with this is the rule of opposites. Boost mids in the vocals and cut them in the guitar, or something similar. Vocals can also have annoying hiss or sibilance, and sometimes cutting high frequencies can help that.
10. Strings, and more specifically good string patches from a synth, usually need little EQ. If they are merely a support player, I may thin them out a tiny bit, or if they are meant to be present, I may thicken them in the mids a little (or sometimes the opposite...this stuff is highly subjective). But they usually work well left alone. Really clean piano or keyboard synth patches are the same way.
11. I like to leave reverb returns alone, but if the reverb becomes annoying and noisy, cutting some high can soften it up a bit...same with strings.
12. Extreme EQ setting create sounds of their own. Experiment. But for a non-novel track, be subtle.
13. AC hum from a track can almost always be fixed by cutting 60 Hz all the way off. (Sometimes this can take away from bass or kick sounds, but I believe that most frequencies audible in a song are above 60 Hz).
14. Play with EQ settings thoroughly to find appropriate settings.
15. I don't mix horns too often, but when I do, I like to leave them alone. Clean horn tracks usually seem fine to me.
16. NEVER EVER EVER force yourself to EQ a track that sounds fine, just because you think you should use the full capabilities of the studio. NEVER NEVER NEVER!

    If anyone out there has rules they use for their mixes, especially for instruments I don't mention or use much, send 'em along. :)

    If you have questions, or have noticed I have left something out, or misspelled, or mis-explained, or (god forbid! hehe) I'm wrong, mail me at **Σφάλμα! Δεν έχει οριστεί σελιδοδείκτης.**or Manny Rettinger at **Σφάλμα! Δεν έχει οριστεί σελιδοδείκτης.**. **Σφάλμα! Δεν έχει οριστεί σελιδοδείκτης.**

A Basic Guide for EQing

by Devin Devore of **Σφάλμα! Δεν έχει οριστεί σελιδοδείκτης.**

Some History

Dating as far back as the 1930's, the equaliser is the oldest and probably the most extensively used signal processing device availible to the recording or sound reinforcement engineer. Today there are many types of equilisers availible, and these vary greatly in sophistication, from the simple bass and treble tone control of the fifties to advanced equipment like the modern multi-band graphic equaliser and the more complex parametric types. Basically, an equaliser consists of a number of electronic filters which allow frequency response of a sound system or signal chain to be altered. Over the past half century, equalisers design has grown increasingly sophisticated. Designs began with the basic 'shelving filter', but have since evolved to meet the requirements of today's audio industry.

Understanding EQ and its Effects on Signals

There are two areas of equalisation that I want to cover. Those two areas are vocals and music. I'd like to discuss the different effects of frequencies within audio signals. What do certain frequencies do for sound and how we understand those sounds. Why are some sound harsh? Why do things sound muddy? Why can't I understand the vocals? I'll try and answer all of these question and hopefully bring some light to the voo-doo world of EQ.

Vocals

Roughly speaking, the speech spectrum may be divided into three main frequency bands corresponding to the speech components known as fundamentals, vowels, and consonants.

Speech fundamentals occur over a fairly limited range between about 125Hz and 250Hz. The fundamental region is important in that it allows us to tell who is speaking, and its clear transmission is therefore essential as far as voice quality is concerned.

Vowels essentially contain the maximum energy and power of the voice, occurring over the range of 350Hz to 2000Hz. Consonants occuring over the range of 1500Hz to 4000Hz contain little energy but are essential to intelligibility.

For example, the frequency range from 63 to 500Hz carries 60% of the power of the voice and yet contributes only 5% to the intelligibility. The 500Hz to 1KHz region produces 35% of the intelligibility, while the range from 1 to 8KHz produces just 5% of the power but 60% of the intelligibilty.

By rolling off the low frequencies and accentuating the range from 1 to 5KHz, the intelligibility and clarity can be improved.

Here are some of the effect EQ can have in regards to intelligibilty. Boosting the low frequencies from 100 to 250Hz makes a vocal boomy or chesty. A cut in the 150 to 500Hz area will make it boxy, hollow, or tubelike. Dips around 500 to 1Khz produce hardness, while peaks about 1 and 3Khz produce a hard metallic nasal quality. Dips around 2 to 5KHz reduce intelligibilty and make vocals woolly and lifeless. Peaks in the 4 to 10KHz produce sibilance and a gritty quality.

Effects of Equalisation on Vocals

For the best control over any audio signal, fully parametric EQ's are the best way to go.

80 to 125
160 to
250
315 to
500
	Sense of power in some outstanding bass singers.
	    Voice fundamentals
	    Important to voice quality

630 to 1K	Important for a natural sound. Too much boost in the 315 to 1K range produces a honky, telephone-like quality.
1.25 to 4K 5 to 8K	Accentuation of vocals Important to vocal intelligibility. Too much boost between 2 and 4KHz can mask certain vocal sounds such as 'm', 'b', 'v'. Too much boost between 1 and 4KHz can produce 'listening fatigue'. Vocals can be highlighted at the 3KHz area and at the same time dipping the instruments at the same frequency. Accentuation of vocals.
	The range from 1.25 to 8K governs the clarity of vocals.
5 to16K	Too much in this area can cause sibilance.

Instruments

Miking instruments is an art ... and equalisers can often times be used to help an engineer get the sound he is looking for. Many instruments have complex sounds with radiating patterns that make it almost impossible to capture when close miking. An equaliser can compensate for these imbalances by accenting some frequencies and rolling off others. The goal is to capture the sounds as natural as possible and use equalisers to strighten out any non-linear qualities to the tones.

Clarity of many instruments can be improved by boosting their harmonics. In fact, the ear in many cases actually fills in hard-to-hear fundamental notes of sounds, provided the harmonics are clear. Drums are one instrument that can be effectively lifted and cleaned up simply by rolling off the bass giving way to more harmonic tones.

Here are a few ideas on what different frequencies do to sounds and their effects on our ears.

31Hz to 50Hz	These frequencies give music a sense of power. If over emphasised they can make things muddy and dull. Will also cloudy up some harmonic content.
80Hz to 125Hz	Too much in this area produces excessive 'boom'.
160Hz to 250Hz	This is the problem area of a lot of mixes. To much of this area can take away from the power of a mix but is still needed for warmth. 160Hz is a pet-peeve frequency of mine. Also, the fundemental of bass guitar and other bass instruments sit here.
300Hz to 500Hz	Fundamentals of string and percussion instruments.
400Hz to	Fundamentals and harmonics of strings, keyboards and percussion. This is probably the most important

1K	area when trying to control or shape to a natural sound. The 'voice' of an instrument is in the mids. To much in this area can make instruments sound horn-like.
800Hz to 4K	This is a good range to accentuate instruments or warm them up. Too much in this area can produce 'listening fatigue'. Boosts in the 1K to 2K range can make instruments sound tinny.
4K to 10K	Accentuation of percussion, cymbals, and snare drum. Playing with 5K makes the overall sound more distant or transparent.
8K to 20K	This area is often what defines the quality of a recording or mix. This area can also help define depth and 'air' to mix. Too much can take away from the natural sense of a mix by becoming shrill and brittle.

Here are a few other pin point frequencies to start with for different instruments. In a live sound situation, I might event pre set the console's eq to these frequencies to help save time once the sound check is under way. These aren't the answers to everything... just a place to start at.

Kick Drum:

Besides the usual cuts in the 200Hz to 400 area, some tighter Q cuts at 160Hz, 800Hz and 1.3k may help. The point of these cuts makes for space for the fundamental tones of a bass guitar or stand up. I have also found a high pass filter at 50Hz will help tighten up the kick along with giving your compressor a signal it can deal with musically. 5K to 7K for snap.

Snare Drum:

The snare drum is an instrument that can really be clouded by having too much low end. Frequencies under about 150Hz are really un-usable for modern mixing styles. I would suggest a high pass filter in this case. Most snares are out front enough so a few cuts might be all that is needed. I like to start with 400Hz, 800Hz, and some 1.3K. This are just frequencies to play with. Doesn't mean you will use all. If the snare is too transparent in the mix but I like the level it is at, a cut at 5K can give it a little more distance and that might mean a little boost at 10K to brighten it up.

High Hats:

High hats have very little low end information. I high pass at 200Hz can clean up a lot of un-usable mud in regards to mic bleed. The mid tones are the most important to a high hat. This will mean the 400Hz to 1K area but I've found the 600Hz to 800Hz area to be the most effective. To brighten up high hats, a shelving filter at 12.5K does nicely.

Toms and Floor Toms:

Again, the focus here is control. Most toms could use a cut in the 300Hz to 800Hz area. And there is nothing real usable under 100Hz for a tom... unless you are going for a special effect. Too much low end cloud up harmonics and the natural tones of the instrument. Think color not big low end.

Over Heads:

In my opinion, drum over heads are the most important mics on a drum kit. They are the ones that really define the sound of the drums. That also give the kit some ambience and space. These mics usually need a cut in the 400Hz area and can use a good rolling off at about 150Hz. Again, they are not used for power.... these mics 'are' the color of your drum sound. Roll off anything that will mask harmonic content or make your drums sound dull. Cuts at 800Hz can bring more focus to these mics and a little boost of a shelving filter at 12.5K can bring some air to the tones as well.

Bass Guitar:

Bass guitar puts out all the frequencies that you really don't want on every other instrument. The clearity of bass is defined a lot at 800Hz. Too much low end can mask the clearity of a bass line. I've heard other say

that the best way to shape the bass tone is to roll off everything below 150Hz, mold the mids into the tone you are looking for, then slowly roll the low end back in until the power and body is there you are looking for. If the bass isn't defined enough, there is probably too much low end and not enough mid range clearity. Think of sounds in a linear fashion, like on a graph. If there is too much bass and no clearity, you would see a bump in the low end masking the top end. The use of EQ can fix those abnormalities.

Guitar/piano/ etc.:

These instruments all have fundamentals in the mid range. Rolling off low end that is not needed or usable is a good idea. Even if you feel you can't really hear the low end, it still is doing something to the mix. Low end on these instruments give what I call support. The tone is in the mids. 400Hz and 800Hz are usually a point of interest as are the upper mids or 1K to 5K. Anything above that just adds brightness. Remember to look at perspective though. Is a kick brighter than a vocal? Is a piano bright than a vocal? Is a cymbal brighter than a vocal?

In Closing

Equalisers are one of the most over looked and mis-used pieces of gear in the audio industry. By understanding equalisers better, an engineer can control and get the results he or she is looking for. The key to EQ'ing is knowing how to get the results you are looking for. Also, knowing if its a mic character or mic placement problem. EQ can't fix everything. It can only change what signal its working with. Equalisers are also a lot more effective taking away things in the signal than replacing what was never there.

Reverb

Reverb is an important studio tool. It can be used to add realistic ambience to a sound that was recorded in a dead, dry room, or to electronic or synth sounds. About everything we hear has some reverb to it, so when we hear an untreated sound, it sounds uncomfortable, and unnatural.

Back in the sixties and seventies before there was digital reverb, studios used plate reverb. They would hang a thin piece of metal inside some frame work, and vibrate it using a voice-coil assembly. Then they would mic the metal plate with contact mics and feed that back into the mixer. The only problem with this method was that it sounded metallic and bright. After so many years of hearing this, people were used to it, and the new digital reverbs sounded strange to them. Now, digital reverb units repeat little fragments of the sound wave thousands of times to recreate reverberation. Most reverb units have hall sounds, room sounds, and, of course, plate sounds which are great for drums.

Basic Rules for Using Reverb

- The effect sounds the best when used in sparingly. Don't swamp tracks in it. Use the least possible to get the desired effect. The best engineers know when they have used too much.
- Sounds with a lot of bass, such as the kick drum or bass guitar are best left with little or no reverb. If you do use it, keep it short and bright, or cut the low frequencies on the reverb return. Otherwise you'll have a big mess before you know it.
- Obviously the more reverb you use, the farther away a sound will seem. This can be used to push certain things back in the mix such as backing vocals, but once again, don't load it on.
- Many times your effects unit will allow you to use many different types of reverb in one mix. Theres nothing wrong with using a couple of different reverb styles all within the same mix, it will just sound more interesting to the ear.

Useful Settings

- **Drums**

  Style:

  Bright Plates, nonlinear

  Length:

  Between 1.1 and 2.5 seconds

  Pre-delay:

  Around 20 milliseconds

- **Vocals**

  Style:

  Plate or short hall

  Length:

  Between 2 and 3 seconds

  Pre-delay:

  Between 20 and 60 milliseconds

- **Piano**

  Style

  Hall or concert hall

  Length:

  Between 2 and 4 seconds

  Pre-delay:

  Between 5 and 50 milliseconds

- **Electric Guitar**

  Style:

Room or Plate

Length:

Between 1.5 and 3.5 seconds

Pre-delay

Between 20 and 50 milliseconds

- **Strings**

    Style

    Plate or Bright hall

    Length:

    Between 1 and 2.5 seconds

    Pre-delay

    Between 20 and 80 milliseconds

10 Steps to a better Mix.

By: Howard Mangrum

The following is a ten step procedure for the mixing of a song. These steps can be varied in any way necessary to accommodate the themes or concepts of the song or materials to be mixed. Please be aware that the detail of each step can change depending on your equipment and the song. Sometime the song may not be fully developed and attempting to mixdown will make this evident, one of the reasons why even the professionals do rough mixes. Final mixes are best approached when your ears are fresh, not at the end of an all day tracking session. After mixing various projects you will develop your own procedure and you can feel free to throw this out, I mean store this for further reference along with all of the other bad song ideas that your friends have come up with.

1. Normalize & Mute

Normalize each track by panning to the center, take the EQ section out or verify all settings are zeroed (this may be in the 12 o'clock position), and turning down (off) all Aux Sends so that there are no effects. Pull all faders down (some people mute each channel, then un-mute them individually as they proceed).

(This is a good place to play a reference CD, to help ensure the monitoring system is performing properly and you have a good referenced starting point for your mix.)

Review any notes taken during the tracking process and your preproduction notes. Setup the signal routing scheme, configure patch bays. Compressors and noise gate can be patched in and normalized so they have little or no effect (put device into bypass if possible, set noise gates to a low threshold, etc.). It should be possible to assign outboard effects (reverbs) to the various tracks at this point based on the song concept and basic ideas of the sonic landscape.

(It is perfectly acceptable to determine the concepts and sonic landscape as you progress through these mixing steps, this is art and there are no rules, just guidelines or opinion.)

2. Loop play

Set the tape deck to play the song in loop-mode if possible. This allows the following steps to be completed in a continuous procession.

3. Critique & EQ

Critique each track individually. Start by soloing (un-muting or only bringing up one fader) each track to ensure proper gain setting by observing your level indicators. Setup noise gates and compressors if necessary. Perform your first rough EQ, do this EQ as fast as possible, don't spend more than a few minutes per track

(the point of dimensioning returns is close at hand during this first pass, the perceived frequency distribution will shift/change, as all tracks are mix together)

(Periodically switch the EQ section out and back in to help ensure you are making improvements.)

(General approach to EQ; if you have a parametric with tunable 'Q' use it to fix frequency problems, get rid of the bad sound. Use shelving EQ's to do gross adjustments to the sound, since they effect a large range of frequencies.)

(It is usually better to cut, so work to cut the bad and if you have remaining control use this to enhance.)

(Adjustment to the EQ/frequency content can make dramatic changes to the gain structure of your signals/sounds, be sure to keep an eye on this and make adjustment accordingly)

4. First Mix

Bring up each track to start building your mix. The order should mimic the priority of each track, this depends on the style of the music and your personal tastes.

(It is standard practice to start with the foundation, such things as drums & bass)

(This is were you start to build your sonic landscape, you determine which sound should be out front and which sound should be in the background.)

(If necessary this is a good place to draw a sketch of your stage setup of the band, to help visualize your sonic landscape)

5. Re-EQ

Re-EQ tracks where necessary. Listen for too much sound (muddy) in each frequency range, where, you have instruments or sounds that are in the same basic frequency range and may conflict or mask each other.

6. Pan

Pan tracks/sounds to complete the setting of your sound stage.

(This step is done in conjunction with re-EQ, step 5. The overlapping frequencies maybe less offending after panning)

(Periodically monitor your constructed sonic landscape in mono to ensure that phase cancellation and sound masking are not going to cause you any problems, your mix should stand-up in mono as well as stereo, with only the basic imaging shifting.)

7. Effects

Setup the reverb and other effects. When applying reverb, keep your sonic landscape in mind. You are setting the outer-boundaries of your sonic landscape at this point.

(It is easy to over use reverb and other effects, generally turn-up the effect to a point where they become dominate then back them off to they just meld into the background)

(Be sure to keep a written record of which effects are used where and the programs of the effect units, with any special settings and/or signal routings that have been employed)

8. Balance Mix

Listen to the mix and ensure you can hear each sound and the over-all balance between each sound is correctly portioned.

(This is a good point to perform a rough mix to tape and playing it on a secondary monitoring system to help gain a second prospective and ensure the main monitoring system is not leading you down the wrong path. Studio monitors can reduce the perceived impact of various settings and the amount of such things as reverb.)

9. Map Moves

Map out any move that maybe necessary, such as:
- level changes
- muting of tracks
- panning
- effect changes

(map the move to a tape counter and/or a smpte time readout, keep a written list of these moves)

(learn to perform the moves on-the-beat, tap your foot and count)

10. Practice Mix

Practice the mix, learn to play the console/mix like an instrument. When you are confident with your mix start recording it to your mixdown deck.

(It is usually good to perform a few mixes, like any performance each will be different and one will usually be preferable.)

(It is common at this point for you to realize that you have not determined how the song should start or end, map these moves out as above. Be sure to allow for some pre-roll & post roll time)

(Performing a mix should be similar to performing on an instrument where moves and other events happen on the beats of the song, your mixing moves should have rhythm to them.)

Roger Nichols Recording Guide

Setup:

- How many people (musicians) will be in the recording room and how will they be arranged
- What Instruments will they be playing and what special requirements need to be met
- How big is the room (or rooms). If sharing an isolation room, consider grouping of instruments for least adverse leakage.
- Isolation between instruments should be considered. Is some of what is being recorded going to be replaced (stand in vocalist, not the real solo, etc.) Determine how best to isolate the instruments (baffles, Tube Traps, blankets, foam, plywood).

Cables:

- You can never have too many cables or adapters. All cables must have previously been ascertained to be in proper working order. Cables that have been previously suspect and checked to find nothing wrong should be labeled as such until they have successfully worked in a session. (this is in case of a cable problem, the first cable to check would be a previously faulty cable.) Anticipate problems as much as possible.

Microphones:

- Choice of microphones. What mics are available for the session? What mics are specifically requested by the client? Are there notes from previous sessions with the same musicians that pointed out a unique requirement or a mic that worked rather well in a particular situation.
- Impedance must be matched. (Lo, Hi, Inline Xformer) Thin sounding microphones (reduced low frequency response) usually means that the impedance is not matched properly. Connections must be matched. (XLR, 1/4", DIN, Teuschel). Polarity must be matched (Pin 2 hot - Pin 3 hot?)
- Phantom requirements must be ascertained. If mics are split to multiple consoles, such as live performances, only one console should provide phantom power. Make sure that mic splitter will pass phantom (some will not). If console does not provide the proper phantom voltage, use external pass through phantom power modules. If the microphone has it's own power supply, make sure that phantom is turned off to that mic, otherwise distortion and noise may result. (Guaranteed in some instances)
- Can't have phantom on with unbalanced microphone.
- Try direct box for synths and electric instruments. Try different direct boxes (they are like microphones and have a coloration of their own). Active (phantom powered) direct boxes may not have ground isolation capabilities and may cause ground loops (which result in a buzz or hum). Some consoles will let you pluginstruments in directly. (check for impedance matching.)
- Pickups on acoustic instrument can be added in with the microphone sound.
- Mic patterns must be chosen properly for the job. Understand proximity effect in microphones.
- Microphone placement may not be the same each time you record in a similar situation. It may depend on the individual player or instrument.
- Listen for reflections off of music stands (use foam or towel) when recording vocals. Listen for extraneous noises from squeeky chairs or rattling instruments.

Speakers and monitoring:

- What do they sound like. Have you heard these speakers before in a different environment? Does the control room color the sound of the speakers so that you must compensate for that difference?
- Placement of speakers in control room may effect the way they sound (experiment with different placements)
- Try not to use speakers (instead of headphones) for monitoring in the studio during recording. The leakage will hamper an otherwise good recording. If this must be done, there are methods whereby two speakers are fed a MONO signal and placed out of phase. The microphone is then placed in the phase null between the speakers. Extreme caution should be taken when employing this method.
- Use distribution system for multiple headphones, don't just parallel a bunch of headphones from the console or cassette machine headphone outputs.
- "More Me" headphone distribution systems for individual mixes to each musician can help the recording process immensely.

Console:

- Trim vs. volume control. Don't clip the input. In some situations it may be desirable to set all faders to "zero" and establish the initial recording level with the input trims. This provides an instant graphical representation if microphone levels change drastically during the recording (the fader is pulled down or up from it's reference)
- Check the sound through the console, select the correctrouting path, with inserts disabled.

- With no EQ, listen. Does it sound good, is it muffled, scratchy, far away,or boomy?
- Is the mic facing the wrong way, (this happens often) or are you listening to wrong input.
- Impedance mismatch between the console input and the source. (line, mic, instrument)
- Bad cord, connection, patch bay, patched in wrong hole, patchbay normal not broken- mic going too many places at once
- Balance vs. unbalanced - pin2 vs. pin 3 (unbal pin 3 at one end + unbal pin 2 = SHORT)
- Bad instrument - change try another
- Bad playing technique or position - try something else, face Mecca.
- Move the mic a little - start with close micing, then move the mic away
- Acoustic guitars, pianos, Bass, Standup bass, Drums
- Go in the room and listen to the instrument with a finger in one ear.
- Ask the player - chances are he has recorded this instrument before and has some idea as a starting place.
- If he says "This is what I do all the time and it always sounded good before" then there is probably something else wrong.

Tape Machine:

- Machine on input. Monitor through the machine (good idea in case you are overloading the machine input) make sure that whole signal path is working right. (what you see on the meter may not be what you think is going there.
- Listen to output of machine with no music playing. Listen for hums, crackles or buzzes. If the meter is reading something, then there is probably a hum or other noise that you didn't notice.
- What kind of metering? Digital metering is the most accurate. Peak meters second best Analog VU meters, depend on what music is playing (click, hi hat, organ, etc.) Percussive instruments should indicate lower on the VU meter for proper recording level.
- Don't forget to make sure that you are using the correct tape for the machine. Bias, tape stiffness & head wrap.
- Noise reduction dbX Dolby A, B, C, S, SR
- Don't use noise reduction on the SMPTE track. Make the SMPTE track one of the edge tracks. (cross-talk). Don't use noise reduction on digital recordings.
- Autolocate is a nice feature. Also autopunch, cycling etc.
- Don't forget to clean the machine. Digital machines need cleaning too. Follow manufacturers guidelines.

Recording:

- Start the machine in plenty of time before the song begins. Allows machine to get up to speed. Allows plenty of SMPTE for future lockups.
- Let the machine keep running a little while after the take. In case you want to add something at the end, or cross-fade into the next tune, or?
- Make sure you have enough tape for the take you are about to record. If what you are recording is longer than a reel of tape, plan a break in the music for changing tape or get a second machine for A/B rolling. (the second machine is placed into record before the first machine runs out of tape.

Overdubs:

- Must be able to monitor output of machine
- Good headphone mix.
- Try not to use speakers to monitor during overdubs
- Test punch in capabilities of machine Punch during sustained playing & punch right on beat. Play back. See if there is glitch and see if there is any delay. If delay, modify punch technique accordingly.
- If big glitch, don't punch during sustains or punch on back beat or someplace that will mask punch.

Effects - EQ:

- Equalizers - change the tonal characteristics of the audio. They have at least bass and treble controls. Most desirable is four band sweepable parametric EQ.
- Graphic equalizer. Usually 5 to 31 frequency bands, each fixed in frequency. Usually with slide pots to show a graphic representation of the frequency curve.
- Peak vs. Shelving EQ.
- Tuning EQ by EAR
- Use EQ to:
  - Compensate for low listening levels
  - Make the blend between different instruments more pleasing
  - Compensate for bad frequency response in some device
  - Reduce noise
  - Special effects like telephone voice
  - Reduce apparent leakage between instruments

Effects - Compressors & Limiters:

- Compressors keep levels more constant by automatically detecting level changes above a set level and riding the gain.
- Use compressors on individual instruments, not mix. It will be less audible.
- Attack time settings determine the "punchiness" of the instrument. Peaks get through before the compressor actually clamps down. Faster attack will make for a smoother sound.
- Limiters are faster than compressors and are there to LIMIT the amount of signal passing. These are usually there to protect equipment such as radio transmitters or speakers from overloading.

Effects - Noise Gate:

- Noise gates work like a soft on-off switch. As the level of the sound gets below a set point, the signal is turned off, blocking any residual noise that may creep through. If not set properly they can be worse than the noise.

Effects - Delays & Echoes:

- A delay by itself has no effect but to delay the signal. When the delay is heard mixed with the original signal, we have a sometimes more interesting sound.
- Echo & reverb units control the amount of feedback sent to the input of the delay as well as the number of taps off of the delay line. These signals mix together to form artificial reverberation as found in different size enclosed spaces.
- Doubling (recording the same instrument playing the same part twice) can be simulated by using a delay of from 9 to 30 milliseconds. This fattens up vocals and instruments and can make it sound like there was more than one instrument playing the same part.
- Short delays can also add fake ambiance to a recording that was too dead sounding.
- Chorusing is caused by modulating the delay time. This modulation causes a change in pitch as well as a change in the delay time. This produces a wavy effect in the sound.
- Flanging effects are created by using a delay of 10 to 20 milliseconds and changing the delay amount slowly between those two parameters. The delayed signal mixes with the original signal and some of the frequencies are out of phase with each other and cancel or augment each other. A change in delay time changes the frequency that is affected.

Harmonizers - Octave dividers - Aural Exciters:

- Harmonizers are used for pitch shifting effects. They can be used to fix bad notes in some cases, or to add harmonies in other cases.
- Octave dividers add an additional tone one to two octaves below the original signal. This can fatten up an otherwise wimpy bass or kick drum.
- Aural Exciters work by adding slight distortion and phase shift to the signal. This can brighten up an otherwise dull sounding instrument. They usually work the best if there is a rich overtone sequence present in the original sound. The work great on snare drums, background vocals, and string pads.

Combining tracks:

- On analog machines do as little as possible. If you have to combine vocals - record a bunch, combine to one track - record next bunch - combine to next track. At all costs avoid bouncing to adjacent tracks (feedback). Watch out for track next to SMPTE.
- Combination digital and analog machine. Record vocals on ADAT & combine to one track on analog deck. Same quality as one original recording on analog machine. Adjacent tracks not a worry on digital machines.

Comping tracks:

- Recording multiple tracks and combining to make one master track. If you can do it on the digital deck, it is better. If you must do it on analog deck, try not to do it multiple times.
- If the way you work is to try 4 takes, then comp, try 4 more then comp again. Don't use the comp track as a component and bounce to new track, try to take any pieces and comp them into the existing comp track. That way comp track will never be more than one generation down. (Make a safety track if you have trouble punching in tight spots.

Mixing:

- Clean up tracks. Erase unwanted material (with the supervision of the producer). Mixing will be easier
- Make a cue sheet reminding you when to make what moves
- Levels. Different DAT machines use different reference level.
- What does reference level mean? analog vs. digital.
- In analog recording, "Zero" is a level reference at which there is 3% harmonic distortion. Above this level there will be more distortion but a better signal to noise ratio. Audio contains peaks which may be above this zero reference by as much as 20dB. Analog tape compresses this information and records it with more harmonic distortion, but for the

small instance that the peak lasts, this may not be a problem. If recordings are made at a lower level, the distortion figures are lower, but the signal is dropping into the noise floor of the tape.

- In digital recording, "Zero" is the level above which no additional information can be recorded. This results in hard clipping of the sound. Anything above "Zero" is not recorded. A reference level of 18dB below "Zero" allows room for peaks in the audio to be recorded without clipping. Because the noise floor is so low in digital (98dB below "Zero") having a reference at -18dB does not really effect the quality of the recording.
- Echo. Don't use too much of a good thing. Use just enough to provide the ambience or effect necessary.
- Effects. If you have empty tracks available on the multi track tape, record the effects to free up equipment for something else, or to save time in re-mixing.
- Limiters (use on record and playback - different ratios) Effects on vocals should be kept to a minimum.
- Panning and stereo placement should be determined by the final destination of the mix (TV, video game, Surround Sound, CD, CD Rom). Keep in mind the center buildup phenomenon. Avoid placing something all the way to one side. (keep in mind stereo listening and being able to hear from opposite side of the room)

Mix machines:

- Analog 2 track
  - Revox, Tascam. Otari, Ampex, Sony, Studer
  - 30 ips vs. 15 ips vs. 7 1/2 ips
  - 1/2 inch vs. 1/4 inch
  - Dolby vs. Non Dolby
  - dbX and other noise reduction.
  - Center track time code
  - Cleaning of machines
- DAT
  - 44.1kHz vs. 48kHz
  - Emphasis on or off
  - External or built in converters
  - Type of DAT tape Computer backup DAT tape, Apogee, HHB.
  - Don't use 3hour tapes unless machine is designed for it
  - Cleaning of machines. Use DAT cleaning tapes properly.
  - Input pause wears the heads.
- CD-R
  - Marantz, Carver, Yamaha, Studer, Phillips, Micromega.
- Cassette
  - No comment
- Mixing back to two tracks of multi-track
  - Multitrack 48kHz or 44.1kHz? Stuck with whatever multi is.
- Digital or analog.
  - Updating mix without remixing

Sample Rate Conversion:

- To get from one sample rate to the other or VSO final mix?
- Alesis AI-1
- Roland SRC-2
- N-Vision
- Z-Sys
- Analog out - in

Editing:

- To change the arrangement of the song
- To assemble all of the tunes in order for distribution or going to mastering
- Razor Blade editing (try to keep blood to a minimum). Razor blade editing can be performed on reel to reel digital tapes under certain circumstances. Special precautions need to be adhered to and a bad edit may not be reparable.
- Hard disk editing Akai, Sound Tools, Sonic Solutions, Turtle Beach Roland, SADiE, RADAR, etc.
- Optical disk editors AKAI, Sony PCM 9000
- DAT editors Sony, Otari, Fostex Music editing vs. assembly editing
- Pause editing DAT is a NO-NO unless plenty of time between cuts.
- Editing for vinyl records (South America etc.)
- Editing for cassette master

Pre-Mastering:

- Assemble in the correct order with proper spacing
- Don't do pause edits on DAT machines unless 5 seconds around edit
- (If that is the only way, let mastering do it)
- Consistent levels (If you don't do it, Mastering will have to)
- EQ All of the selections should have similar tonal quality
- When you are done, Make a Digital Copy. Don't send your only tape
- Some plants can accept CD-R as master. It must not be Multi-Session
- All Plants accept Sony 1630
- Some plant will accept DAT(not if they have to edit)
- Include accurate timing sheet (where you want each cut to start
- Make them send you a ref (plant or mastering facility)
- If everything done (eq, levels, editing) copy DAT to 1630
- PQ codes on tape? or PQ time sheet. Music @ 3:00 into tape
- If you can afford it, good idea to let mastering facility EQ and level correct your tape. You want your product to be competitive with everyone else so it has to sound as good. Third party reference is good.
- Think about breaks for cassette. Second side should be shortest

Labels:

- Multi track labeling of boxes and track sheets.
- DAT labels & J cards
- Cassette
- CD labels.

Keeping notes:

- Keeping good notes. Which mic on which instrument.
- Which sequence was used to print to tape
- What was the tempo
- Which SMPTE interface was used to drive sequencer
- What kind of direct box was used through what preamp?
- Was instrument delayed? if so, which delay and by how much?
- What kind of tape was used and what was the machine set up for
- What was the reference level for recording.
- What reference tape was used to set up machine
- What reference tape and levels were used for Mix?
- What effect units were used and what were the settings?
- Limiter settings for vocals or whatever?
- If you printed alternate mixes, what were the differences?
- Were they printed at different levels or different VSO settings?