Production Training

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Spot Creation Process - An Overview

A radio spot can be anything from a 30 second sponsorship announcement for a client, a sting for a specific show, an arts drop, a station promo (which is a message from the station itself such as competitions, FBi gigs and the like) or a show promo.

Usually the process of creating a spot goes something like this:

- Get the brief. This will usually be a piece of copy, perhaps with specified music.
- Read the brief and get a handle on the concept. This is the point at which to ask questions. You want to make sure you know exactly what is required so you don't have to redo work later, make sure you know:
    - Who the spot is aimed at?
    - What kind of vocal is required – what kind of tone of read?
    - What are the music requirements?
    - When does it have to be on air?
    - Does the copy all make sense to you?
    - Do you know how to pronounce all the names/words in the copy.
    - When the spot is going to be scheduled. Is it only going in the dance shows, or run of station, or only going to be played once, or going to be on air for a long time?
    - Who is going to do the scheduling?
- Source the music – covered in chapter 8 – 'Building a Spot'.
- Record the voice-over – covered in chapter 7 – 'Vocal Production'.
- Edit together the spot – covered in chapter 8 – 'Building a Spot'.
- Mix the spot – covered in chapter 10 – 'Mixing'.
- Clear the spot with clients/FBi Staff.
- Load the spot onto the DRS – covered in the document: 'HOW TO LOAD FILES ONTO THE DRS'. Which is on the CD you receive with these notes and also in the root directory of the audio server.
- Make sure the scheduling is taken care of. This means ensuring that either Dan Conway, Tim Boffa, or Jacinta knows that the spot is on the system, what the spot is named and where it is, and what the scheduling requirements for it are.

You may not be involved in all these steps – but if you are involved in this process at all – as a producer it is your responsibility to ensure that all the steps are taken – either by you or some-one else. Do so. Make sure it all runs smoothly by talking to the right people and following the process through to the end.

Workflow

In order to be fast and efficient, it is important that you master your workflow – in other words, all the technical steps and procedures required in order to finish a piece. I cannot stress enough the need to be organised and to either follow existing, or make your own, conventions for file naming, file management etc... If you let this get out of hand – you can at times find yourself spending enormous amounts of time trying to find files, wandering back and forth, figuring out which was the last mix etc...

Anatomy of a Pro-tools Session

A typical Pro-tools session will look like this:

When you create a session, Pro-tools creates a folder and names it whatever you've called the session. This is the 'session folder'. Within that folder it then creates the following:

- A session file named the same thing as the folder. This is the 'map' of the audio you are cutting up. This file contains all the information about your edits, mix, plug in settings etc... In short, everything except the audio itself.
- A folder called 'Audio Files' – this is where all the actual audio generated in the session are stored.
- A folder called 'Fade Files' – this is where all the fades generated in the session are stored.

Note: You may have more than one session file. If you 'save as' you will generate a new session file. This does not generate any new audio data, only a new session file which is quite small. Don't be afraid of using this function – it's invaluable and generates only a tiny amount of new data. This new session file will still refer to the same audio in the 'Audio' folder. When doing a 'save as' – **MAKE SURE THAT IT IS SAVED IN THE CORRECT SESSION FOLDER RIGHT NEXT TO YOUR OTHER SESSION FILES!** Saving a session in the wrong place is an easy mistake to make and can cause you endless headaches later on – including losing a whole bunch of work. This also applies to whenever you are asked to specify where a file goes – always double-check to make sure you are putting it into the right place!

Cool trick

Often you will want to store files or folder in such a way that the file names include dates, and that can be sorted by these filenames. Computers (stupid things that they are) will misread standard date formats 06-04-04 and get the order all wrong.

So... If you want your folder and files organised by dates use the following format: yymmdd.
So first the year, then the month, then the day. For example:

16th July 2003 would be 030716.
28th of Feb 2005 would be 050228.

Either use this at the very beginning of the file name or as the first thing only after something that is the same for all the file names you wish to sort. As in:

Demo_spot_030716_the_spazzys
Demo_spot_030722_radiohead
Demo_spot_030804_1200_techniques
Etc...

Audio File Types

There are myriad different types of audio files out there in the world. Here are descriptions of the ones that you will most commonly encounter at FBi:

- Wavs. Usually named 'filename.wav'. These are mostly used on PC but are a becoming an increasingly universal standard. In order to get a wav file into 'Pro-tools Free' you have to use the 'covert and import' function.
- Aifs. Usually named 'filename.aif'. These are standard Mac audio files and are the file type that Pro-tools automatically generates. Use these whenever you can.
- SDII. Usually named 'filename.sd2'. Called Sound Designer II. These are an older Mac format and are becoming more and more rare. Avoid.
- MP3. Usually named 'filename.mp3'. Technically known as 'Mpeg Layer3', Mp3 files are generated using data compression, which means that they are lower sound quality than CD but they are also MUCH smaller files. How big exactly depends on the bit rate the mp3 was recorded at which can vary from 32kbps (which is really crap sounding – kinda like a telephone) to 128kbps (pretty stock standard – most internet mp3's are this rate – they sound O.K but not fantastic) or right up to 320kbps (which sound pretty good – not too much worse than CD.)

File Types in use at FBi

The Audio server uses .wav files.
So when you finish your piece, bounces should always be made to .wav files.

For Pro-tools sessions use always .aif files.

Always use 16bit 44.1Khz audio.

Anyone generating finished products in mono will have fish thrown at them.

Import Angst and Missing Audio

When you import audio into your session (for example adding a top and tail to your piece) Pro-tools will sometimes make a copy of the audio and place it in your 'Audio' Folder. And under other circumstance it will not – and will instead just refer to the original wherever it may be.

If Pro-tools has to convert the file that you're importing into a different format in order to import it into your session – it will generate a new copy and ask you where to place this copy (in this case stick it into your 'Audio' Folder). If it doesn't have to convert the audio it will (usually) leave the audio where it is and just read it from where it is as you work.

Pro-tools does this in order to save space by not making duplicates of audio unnecessarily.

This, however can be a major hassle since if the audio you are referring to is moved, or renamed, then Pro-tools will not be able to find the audio referred to in your session and suddenly your session will be missing bits. Alternatively, if you move your session folder to elsewhere on the network – again, Pro-tools may not be able to find some of the bits of audio that make up your piece. This is bad.

The best way to avoid this happening is to copy the files you need, manually, directly into the Audio folder of the session your working on, then from within Pro-tools, import the files from your own audio folder.

This means that regardless of where you move your session data – it is self contained and will open correctly with it's relevant bits.

Name Everything!

Get into the habit of naming all audio files as soon as you record them. Especially when you've recorded a long vocal or two different vocals, and then start cutting them up – suddenly you've got lots of generically named vocal regions and can't find the bit you need. Yuk!

Also apply any audio-suite effects such as 'gain' to the whole file, again, before editing.

Naming everything applies to everything you touch – NEVER let a burnt CD leave your possession without being clearly labeled (including what format it is: audio/PC/Mac etc...) Name all bounces, label your computer monitor with the label maker – Just name everything!

When naming, work out some naming conventions, such as naming all vocal tracks 'V-Phillip, or 'V-Levans'. And perhaps all music something like 'M-Grinspoon'. This means that your region bin will end up being nicely organised and you will be able to find things real quick.

Also get to know the naming conventions FBi uses for files on the Audio server. If you name your files according to this protocol as you bounce them – it means you won't have to rename them later and then have two differently named versions of the same file floating around. The 'NAMING CONVENTIONS FOR DRS FILE TYPES' document can be found on the CD you receive with these notes, and also in the root directory of the Audio Server itself.

Cool trick

You can turn off the display of names in the audio regions of the edit page. Sometimes when your editing these can really get in the way of seeing what's going on.

Look under the Display Menu.

And while you're there, you probably want to turn off the 'Display Auto-Created Regions' thing. Have a look at the manual and figure out what this one does...

Further Workflow Ideas

Here are some further tips for creating an environment for yourself that allows as much of your time to be spent creatively as opposed to chasing your own tail:

- When setting up your session always have one audio element per track. If you have a piece of vocal and then later on some music on the same track, or even vocals by two different people – this will seriously hamper your mixing process later on.
- Get used to minimising screen clutter as soon as you create a session. You don't need to see the IO column in the edit page. You don't need to see the Beats and Bars, Samples or Tempo rulers etc... all you need up there in the ruler is the time and markers. Increasing the working screen size as much as you can will make working with Pro-tools a significantly more pleasant experience. Figure out what all that stuff is on screen and how to make it go away.
- Use the 'Narrow Mix Window' mode... aaah!
- Start your piece at 1 minute. Leave some room in front for playing around or moving files when you don't need them. I tend to start a piece at 1 minute and drag lots of audio after the piece. Leaving yourself a mess at the end of your timeline can be great for suddenly discovering weird combinations of tracks that you would never have thought of using before. Leave yourself some mess in order to open yourself to the possibility of magic.
- ALWAYS create a stereo master fader and have it visible one side of the screen.
- Always make sure your computer's clock and date is set correctly. Being able to find files by 'most recently modified' is an invaluable tool, and if your date or clock was out for a day or two – it all goes bad.
- Create a Library. In your folder, create a place where you can store; some sound effects, some pieces of music you want to build stuff out of, your favourite FBi Vocal tags, whatever. Developing your own resources to work with is an important part of developing your 'sound'. And organise this library to make sure it's easy to navigate, a folder for music, a folder for vocal bits, etc...

Cool Trick

File names beginning with a space " ".

When organising your own work folder, and creating sub-folders, it can be useful to have certain folders, the ones you use most often, at the very top of the list. Not easy when things are ordered alphabetically you say – hah!

Naming a folder with a space before the name will mean that this folder is sorted, alphabetically before all other folders.

Neat huh?

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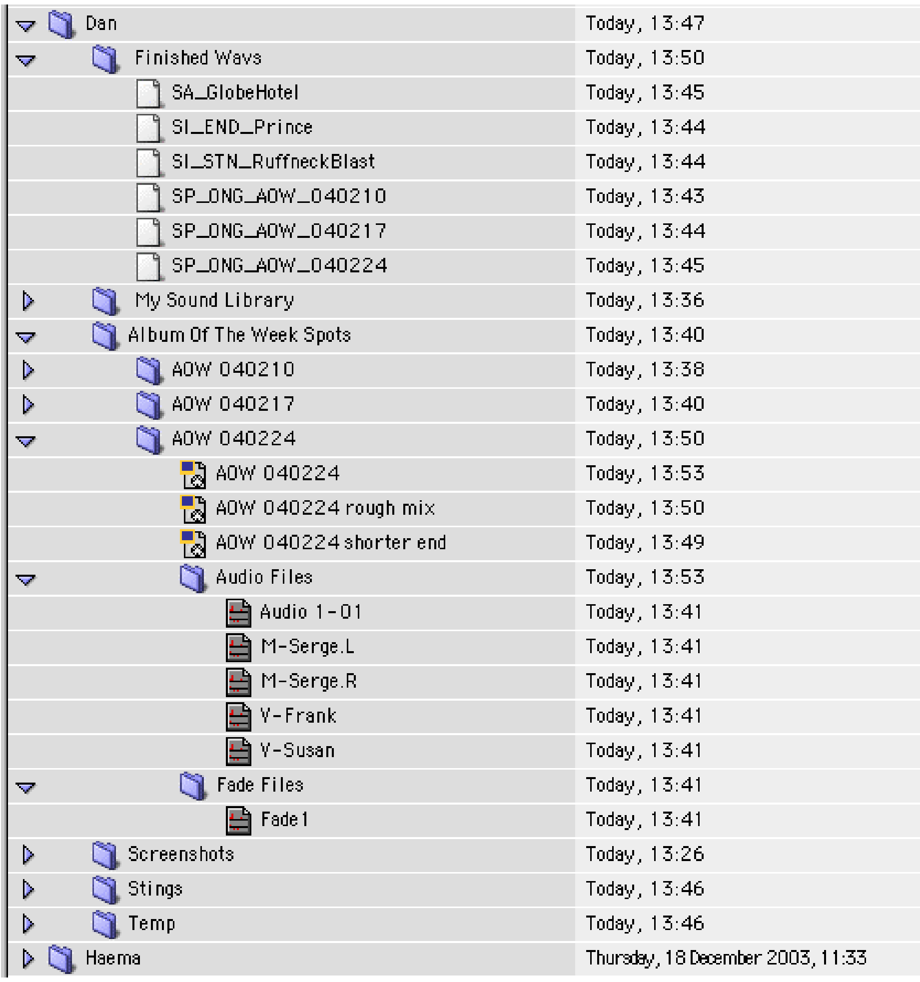
- Archive your work. Make sure you periodically make copies of all your finished work and give them to your Gran for safekeeping. Hard drives crash, folders can be deleted accidentally, bags get stolen, and one-day you'll want to use all this work to either score a cool new job, or bore the grandchildren with.
- Clean up after yourself. Make sure you never leave files lying around the network, or a workstation. Keep all your stuff either in your own folder or as finished product on the audio server. Data left elsewhere **will** be deleted – and it's usually just when you were thinking of going back and using it again.

So, once you really set up to rock and roll and have the world at your fingertips, your folder structure may end up looking something like this...



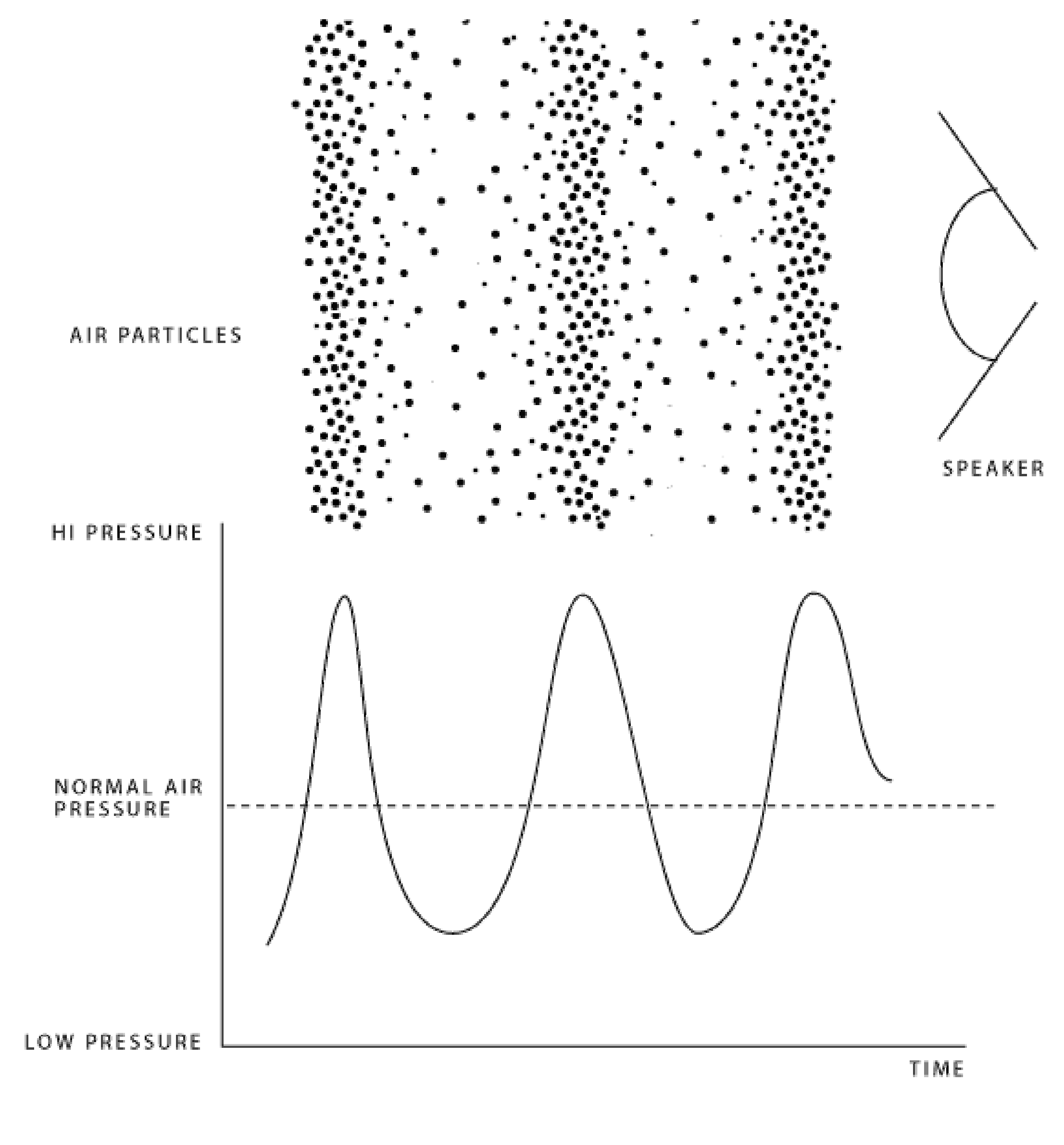
Note that the "Finished Wavs' and 'My Sound Library' Folders are presented at the top of the list because they have a space at the beginning of the folder name. Note also the naming convention used in the AOW spots to order them according to date and the naming convention in the audio folder (these will have been named within Pro-tools) to distinguish Vocal and Music files.

The Fundamentals

Sound, Waves, Propagation, Frequency, Amplitude.

Sound is:
The compression and rarefaction of air.

This means the squashing together and pulling apart of the particles that make up the air around us. This results is waves of pressure. These pressure waves move through the air and our ears interpret these as ‘sound’.



SOUND PRESSURE RELATED TO AUDIO WAVE FORMS

This is what a sound wave in the air really looks like: Areas where there is a higher density of air, and areas of a lower density of air – the waves of pressure here is what makes up sound. These pressure waves can be generated by any moving, vibrating surface – ranging from a human voice box to a piece of cardboard being moved around by a magnet (a speaker).

You can graph the pressure of air as it changes over time. This results in what you will recognise as a waveform. That is what a waveform is – the graph of air pressure AND ALSO – the movement required of a diaphragm to make that wave.

So...

A speaker will follow the exact motion of our waveform (or graph) in order to generate the air pressure waves that we recognise as sound.

Geddit?

Frequency:
How often the graph moves across the center line in a given time unit is the frequency of the sound. We measure it in Hertz – which is Wavelengths per second. If our wave moving only slowly across the centerline of our graph (or the speaker is only flapping back and forth slowly) these are low frequency waves – which we hear as bass or bottom end. If our wave is moving very quickly across the centerline of the graph (our speaker is moving very quickly back and forth) these are high frequencies – which we perceive as treble, or tops.

In nature you will almost never get a simple wave like this. Complex waves – are made out of lots of different frequencies present at the same time – the speaker (and our air) can move with all these different frequencies at the same time. This is also known as modulation.

Amplitude:
How much pressure exists in the air – or how much the speaker has to move, or how high our waveform is – is called amplitude. This equates roughly to volume. High amplitude (big pressure waves, speaker really pumping back and forth) is loud, low amplitude (low pressure waves – speaker just moving a bit) is quiet. Hence it takes more energy to generate higher amplitudes.

Sound Characteristics

Sound travels at (about) 343 metres per second in air. It can travel faster in denser mediums such as water or faster still in things like steel.

Sound, like water waves, bounces off objects (reflection), has some energy absorbed by objects (absorption), it can bend around objects (refraction) and can be scattered by objects (diffusion). How much of each it does depends mostly on the what the object is made of. Some materials (like glass or tile) absorb very little and reflect a lot (hence echo-ey bathrooms). Some materials like fabrics and your Doona, absorb a lot and reflect very little (hence the nice quiet feeling in your padded cell).

Imagine dropping a single pebble into a pool of water and watching the resulting waves. Imagine what happens when they reach the edge of the pool and bounce back and start interacting with other waves still coming the other way. Every time two peaks meet, you get an increase in the height of the wave, every time two troughs meet, you get a deepening of the trough. Every wave is reacting with every other wave it meets.

Imagine these different scenarios for dropping your pebble in our pool of water:

- In the center of a round pool of water
- Towards the edge of a round pool of water
- In the center of a square pool of water
- Towards the edge of a asymmetric pool of water.

Gets complex fast huh?

Cool Trick

Lightning Calculator.
Every seen lightning strike and wondered how far away it was from you?
Here's a way to find out...

When you see a flash of lightning – start counting seconds. 'One cat and dog, two cat and dog' etc... Time how long it takes for the sound of the lightning to get to you. Since sound travels at 343 metres per second – every 3 seconds between seeing the flash of lightning and hearing the thunder represents about a kilometre that the sound has traveled.
So a six second pause means the lightning was about 2 kilometres away from you.

Audio File Attributes

There are a bunch of different variables that apply to audio file types. You should always be aware of the nature of the audio you are working with. Here are some of the key attributes you should be aware of:

Sample Rate
This is how many samples there are per second. Higher numbers here mean higher sound quality audio, but also larger file sizes. CD's have a 44.1kHz sample rate. This is the sample rate you should ALWAYS be working in. If you choose 48kHz, or something else you will have to convert somewhere along the way – which is annoying, and potentially will degrade your sound quality.

Bit Depth
This means for each sample, how accurate the measurement is. 16 bit means that every sample is represented by a number of up to16 bits in size (a number between 0 and 65536). 24 bit means that every sample is represented by a number of up to 24 bits in size (a number between 0 and 16777216). The higher the bit depth the more accurate the sound is, but again, the larger the file will be. ALWAYS use 16 bit settings on any gear you use at FBi.

Mono/ Stereo
Does the file contain one signal or two? 'Stereo Interleaved' means that the two signals (left and right audio) have been combined into a single file.

Cool trick

A few quick ways to work out how large an audio file should be.

1 mono minute of 44.1 kHz 16 bit Audio (CD quality) = (roughly) 5 Mb.

i.e: a three minute stereo track will be (roughly)
3 (minutes) x 2 (tracks) x 5 (Mb) = 30Mb

For MP3's:

A stereo minute at 128kbps = (roughly) 1 Mb
A stereo minute at 320kbps = (roughly) 2 Mb

Frequency and Equalisation

Equalisation (or 'Eq') is really about tone control. It's about shaping the tone/colour of a sound. This is achieved by reducing (cutting) or increasing (boosting) some frequencies or a range of frequencies that make up the sound.

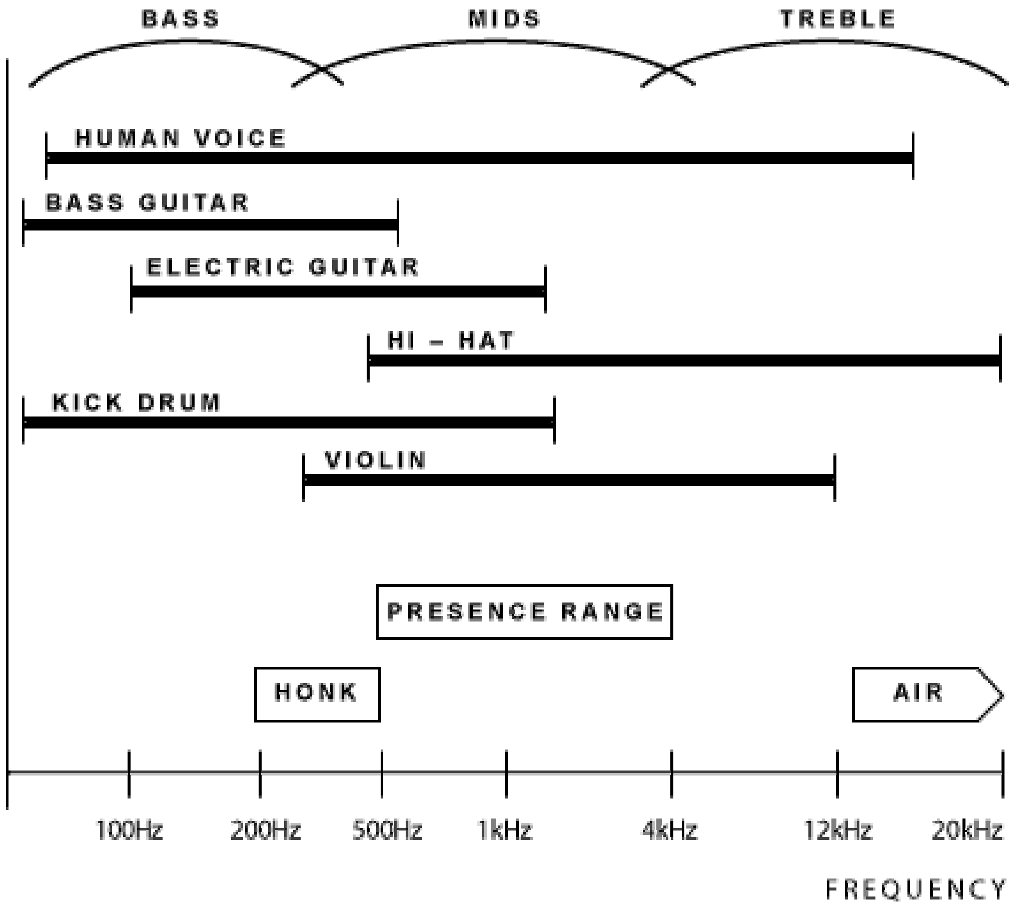
The same type of process is used on your car stereo with it's tone controls – bass and treble, your home stereo with it's 5 band graphic equaliser, or in a studio with parametric EQ.

The human hearing range stretches from 20 Hz (a waveform of 20 cycles a second) to 20,000 Hz. 20,000 Hz can also be expressed as 20KHz (Kilohertz), or sometimes 20K.

Reminder

Lower Hz = slower cycles = low frequencies, or bass, or bottom end.
Higher Hertz = high frequencies, or treble, or top end, or tops.

Any sound occurring in the natural world has frequencies occurring over a big part of the frequency range. The human voice, when mic'ed closely will have frequencies present from about 200 Hz right up to16 K and beyond (female) and 100 Hz to 16K (male). With musical instruments the lowest major frequency is called the 'fundamental' and is the frequency of the actual note of the instrument – above that there will be 'harmonics' which are mathematically related to the fundamental, and give the sound it's character.



FREQUENCY SPECTRUM

Equalisation is used to select and either diminish or increase certain parts of a sound's natural frequency range. It can be used 'technically' – to make something sound 'better' or to reduce unwanted elements of the sound, or it can be used creatively – to make things sound different – transform sound.

Terminology

Overtones – another word for harmonics.
Odd Order Harmonics – the fundamental freq multiplied by 1,3,5 etc... Lots of these present tend to result in a nasally or hollow sound: oboe, ride cymbals
Even Order Harmonics – the fundamental freq multiplied by 2,4,6,8 etc... Lots of these present tend to result in a warm, or full sound, piano, crash cymbals.
Rich – lots of nice harmonics, particularly in the mid and top end
Thin – not much bass, or fundamental.
Toppy – more high frequencies than bass.
Presence Range – the range of frequencies the human ear is particularly sensitive to, and where most of the information of human speech is: 1000-4000 Hertz.
Warm – more bass than high frequencies.
Bright – lots of treble, or high frequencies present.
Dull – not much treble, or high frequencies present.
Boxy – frequencies around 300-500 Hz present.
Flat – no equalisation applied – the original sound (at least in terms of frequency).
Cut – reduce a certain frequency or frequency range.
Boost – increase a certain frequency or frequency range.
Roll Off – reduce all frequencies above or below a certain point.
Attenuate – to reduce (can also apply to level or volume).

Each individual instrument has a particular frequency range – many quite broad. Adding a bunch of instruments or elements together and for them not to clash, often requires you to remove, or reduce certain frequencies where instruments overlap each other in order to gain clarity. This is a big part of what mixing is about – sculpting your sounds so that they fit together and don't crowd each other out, turn into a big mess, or get lost in or behind other instruments.

Each element in a mix should have it's own little place in the frequency range. By the time a final mix is complete – as in recorded music – there will usually be frequencies, and elements, in pretty much all of the audible spectrum.

Types of Equalisers

The Tone Control on your grandma's wireless – all this does is attenuate (or turn down) the top end – changing the balance of the sound. This is a type of equaliser – but about as primitive as you get.

The bass and treble controls in your car is a of type equaliser that boosts everything below or above a certain point (about as subtle as a rock through a Rembrandt).

Your home stereo 5-band Eq offers a little more control in that you can cut or boost certain frequency ranges (starting to become useful).

30 band Graphic Equalisers – use the same principles as your home stereo graphic Eq, but with heaps more frequencies – therefore much more control. Most often found in live applications – behind the mixing desk at live gigs for example.

A Parametric Eq allows you not only to cut or boost frequencies, but also to decide where those frequencies are – kind of like being able to grab one of those faders on your stereo and move it from side to side – from bass to treble – to allow you to find the bit of sound you wish to cut or boost. Proper parametrics also have an extra control called '**Q**' controlling how wide the range of your cut and boost will be. This allows you to create extremely narrow little peaks or troughs - good for isolating a specific frequency for some fine surgery, or a quite wide frequency band to cut and boost, for either radical effects, or when used gently, for more subtle manipulation of the sound colour. Q can also be known as bandwidth.

In a studio, or in Pro-tools, you will most often find 'parametric equalisers'. These are simply a more powerful, flexible and 'surgical' way of controlling the frequencies, or tone colour, of a sound. These are your primary tools in the world of audio surgery – get to know how to use them instinctively.

Filters – These cut everything above or below a certain frequency – the slope of the filter is usually set and is usually pretty steep – kind of like lopping off a whole portion of the spectrum. Useful for getting rid of bottom end on things that don't really need it – or effects such as the telephone voice – see below.

Shelving EQ's reduce everything above or below a certain frequency, but also allow you to control how much reduction is applied – so can be used for more gentle and natural sounding sculpting. These also have a side-effect of boosting the frequency around where you cut – which is an interesting aural effect and can be very useful and musical if used in the right way.

Experiment and compare filters and shelving Eq's and think about what kind of applications they would be good for.

At the end of this chapter you will find some diagrams of the kind of shapes Eq's can be used to create in your frequency range.

Energy

One of the characteristics of frequency in sound is that it takes much more energy to produce or reproduce low frequencies than high frequencies. Sound systems generally use massive amplifiers for each bottom end speaker (coz they have to move lots of air back and forth) whereas one much smaller amp can run the whole top end of a sound system. Same with speakers and also in the real world – It just takes more physical energy to move the big chunks of air around that are bass.

This characteristic of sound is really important – coz in mixing – there's only so much energy to go round. A hi-hat sound (with practically no low end energy) can sound really loud while barely registering on the meters. A bass sound can be almost off the scale of the meters and not sound that loud.

This means that using the bottom end of things that don't really need them is a waste of energy and means your overall mix won't be as loud... Only use bottom end when you really have to. The less bottom end your audio has the louder it can be on-air – hence single mixes often aren't really designed for clubs and vice versa.

Cool Trick

To really cut through on radio and be LOUD in the mix – you cannot go too treble!!! Try making a 'telephone' voice by putting in a Hi Pass Filter and Low Pass Filter, moving them toward the center until you only have a narrow band left at around 1K.

This kind of bright, narrow sound works really well on radio and can really punch through a mix – but it can also sound extremely irritating and harsh – be careful.

It is useful when mixing, particularly in a busy mix with lots of sounds, to restrict each instrument to only the frequency bands where they are most useful and have the most information – often you will be able to cut a lot of the bottom end out of instruments without losing to much 'information'. And think about the spectrum – you want something present in each frequency. Make a plan – what goes where?

The human ear also doesn't really like a lot of boosting – it can sound unnatural and tiring on the ear – so it's better to cut what you don't want than turn up what you do.

Examples of EQ usage:
- To reduce the rumble in a vocal recording (air conditioner, street noise etc...)
- To reduce the boxiness or honk from a vocal recording done in a small or untreated room (go explore the 200-600 Hz frequency range for this one – often a really murky, un-useful area).
- To bring out the most important or musical frequencies of a sound.
- To change the tone colour to fit into your overall mix of elements.
- To cut frequencies out of a sound where you don't want it to interfere with another element – useful for 'nesting' sounds – keep reading for more info on this trick.

Cool Trick

A good technique for using Parametric Eq is to – Find it and Kill it!

Dial in a fair degree of boost – 6-9dB or so – then sweep this peak up and down the spectrum until you have found the bit of sound that really annoys you. This could be a honkyness in the lower-mids, a rumble in the bottom end, some annoying background noise or hiss, whatever. Once you've found this, cut it by reducing the gain there by however much you think is necessary.

Radio Vocal Sounds

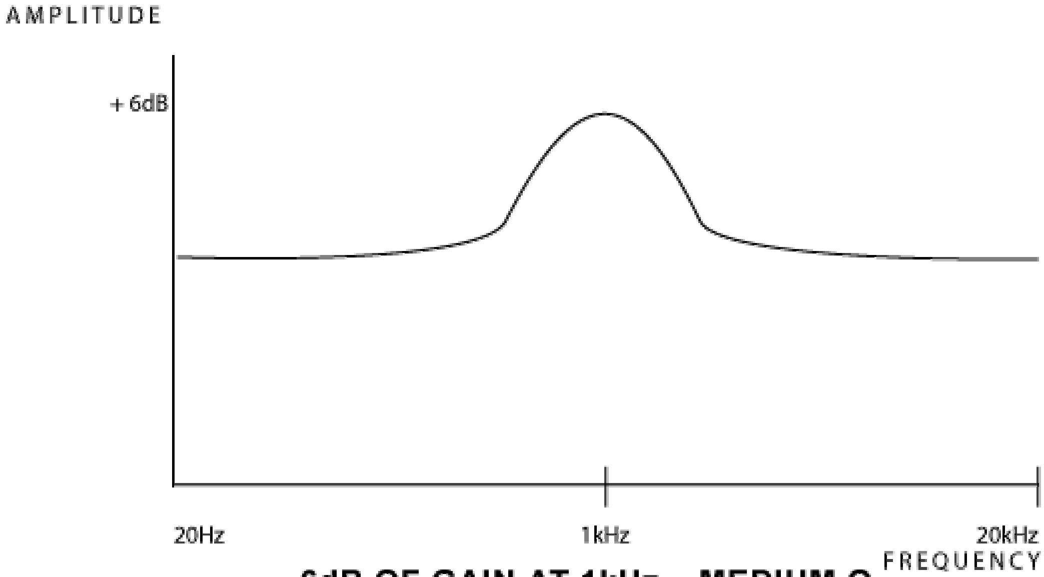
When I EQ voice for radio I will typically dial in a healthy boost in the top end – up to 5-7 db of shelving eq at around 4-6 K. When I want warmth I will also boost the bottom end – somewhere around 100-300 Hz, and often remove any gunk that's cludging up the lower mids (300-500 Hz) with a narrow little cut. I will then apply compression and often limiting as well – this will be covered later in the course.

Cool Trick

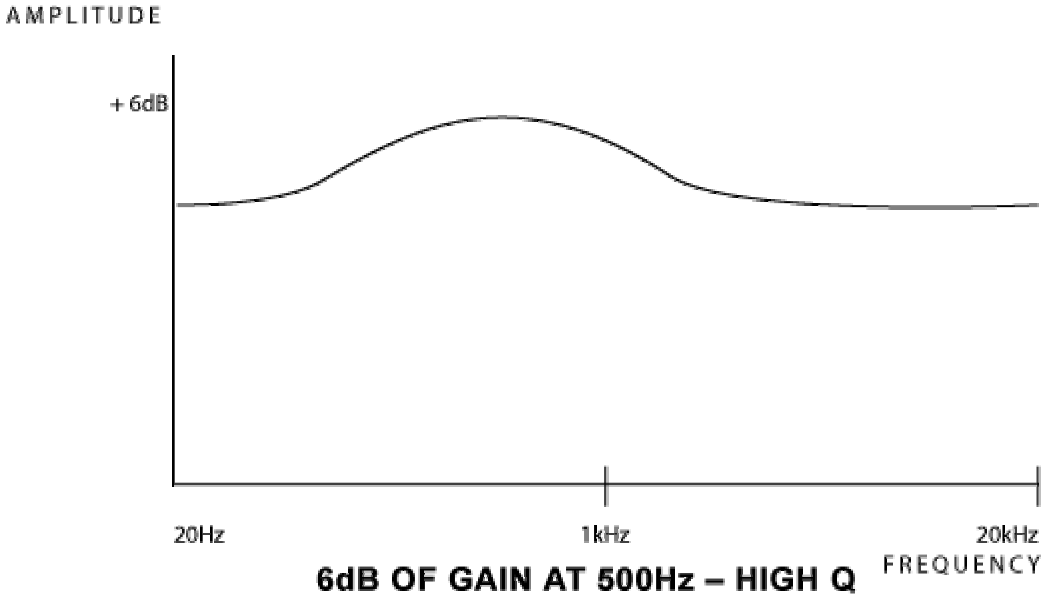
Nesting Sounds.

If you have a vocal that is being swamped by music – Use a parametric Eq on the vocal and find out where most of the information is. Do this it by creating a 6db boost, and then, while the music track is playing, sweep your peak up and down until you find where the vocal stands out the most. Leave a bit of a boost there on the vocal and then apply a parametric Eq to the Music Track(s) and cut a reasonably broad curve (not too much; 4-6dB should do) at that frequency and... Presto! Suddenly the vocal sound is sitting better in the mix and not being stomped on so much.

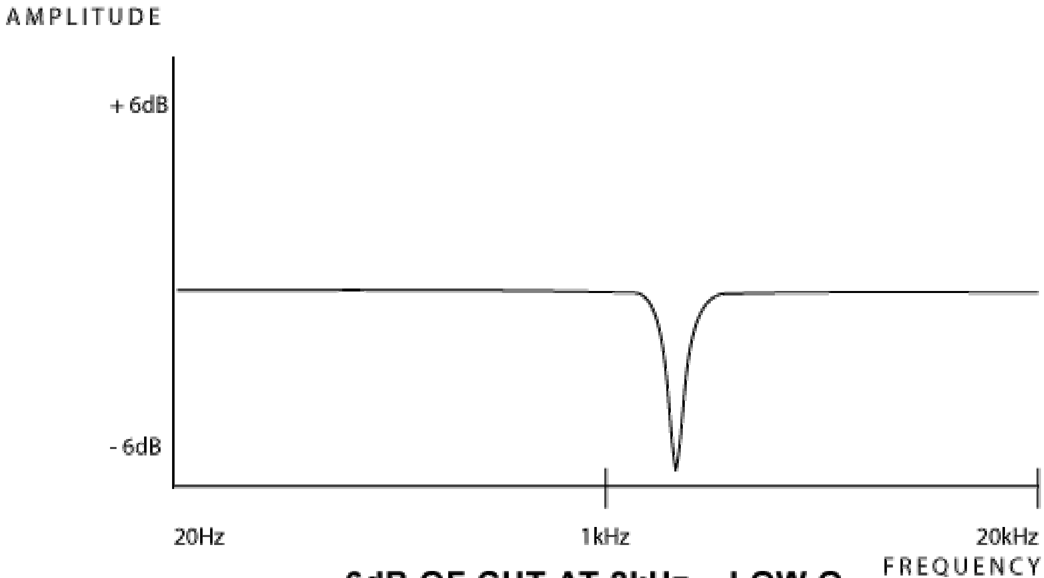
Examples of Eq Curves



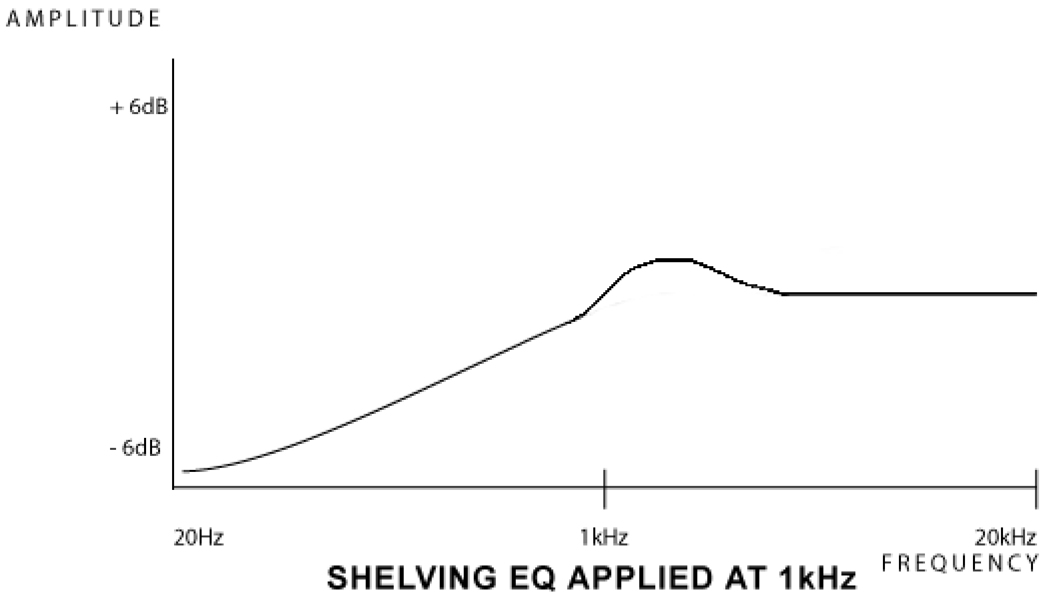
6dB OF GAIN AT 1kHz – MEDIUM Q



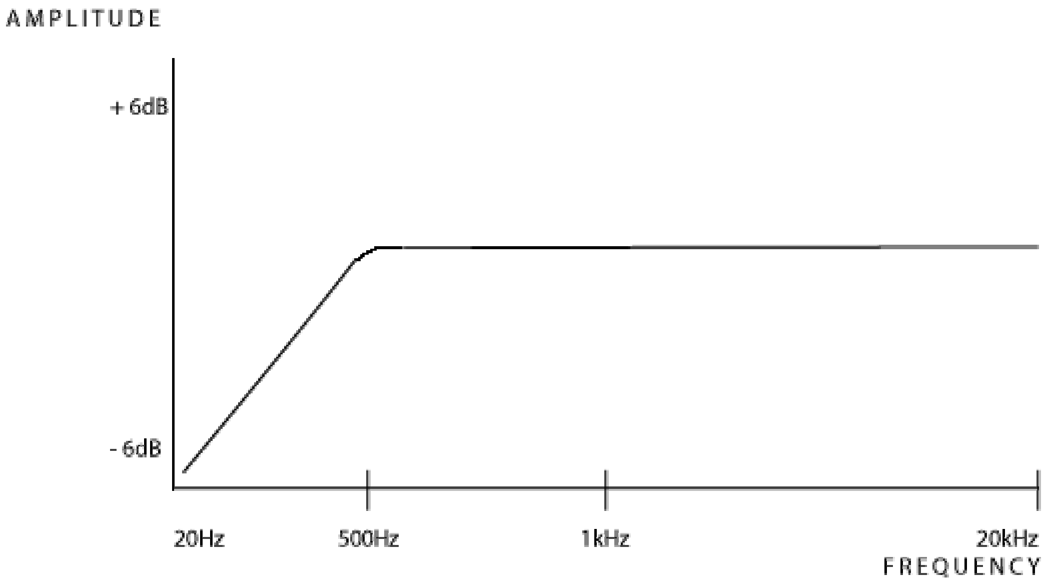
6dB OF GAIN AT 500Hz – HIGH Q



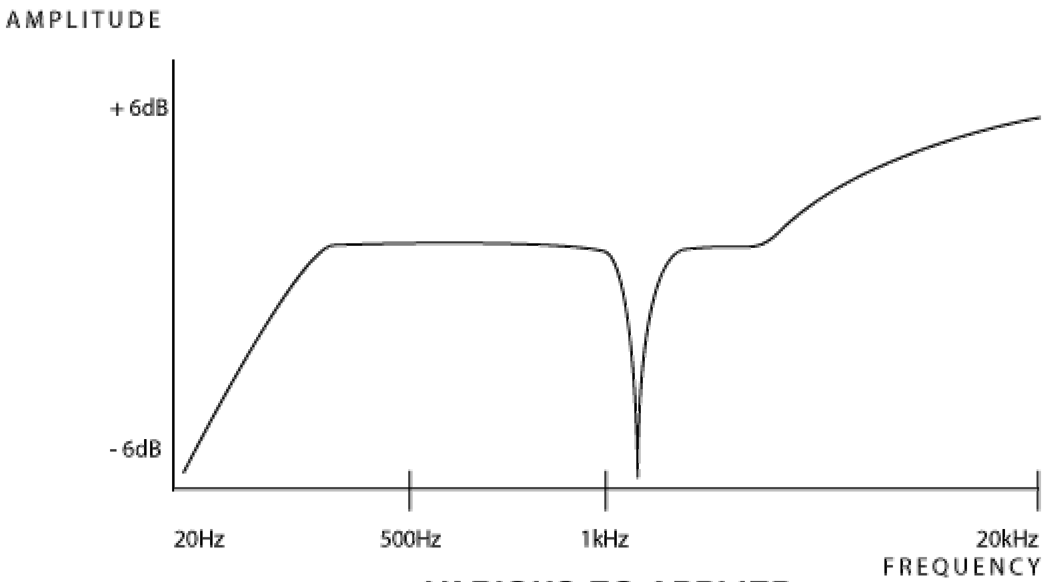
6dB OF CUT AT 2kHz – LOW Q



SHELVING EQ APPLIED AT 1kHz



HIGH PASS FILTER APPLIED AT 500Hz



VARIOUS EQ APPLIED

Dynamics and Dynamics Processing

Dynamics processing is really just automatic level control. It's a process where a piece of equipment (or plug-in) senses the level of the incoming signal and then turns it down (or up) according to the controls offered by the equipment and what you've set them to do.

The dynamics, or dynamic range, of a piece of audio is the difference between the loudest sound and the quietest sound present. So dynamics processing is all about changing the **range** of levels present in your audio.

Sometimes it helps to imagine a little gremlin in a box with a big volume pot who is carefully listening to the signal coming into the box. The Gremlin adjusts his volume control and thereby changes the level of the outgoing signal according to the instructions you've given him.

There are 5 main types of dynamics processing and they all do slightly different, but related, things.

Compression
A compressor will reduce the level of the incoming signal. At what point gain reduction is applied is determined by where you set the 'threshold' control. And how much gain reduction is applied is determined by the 'ratio' setting – which controls the ratio of incoming to outgoing signal. Compression is used commonly on vocals, and also on a whole mix and is applied again at the transmitter to iron out the signal some more before broadcast.

> **Multiband Compression**
> Another form of compression, this is a process where the frequency spectrum is divided into several sections (or frequency bands) and these are all compressed separately before being added together again. This allows much more control over the signal and is sometimes used in transmission but rarely in the production process.

Limiting
A limiter is similar in function to a compressor but instead of being able to select what ratio, or how intense the gain reduction is, it simply prevents the signal from getting any louder at all beyond the threshold level. Limiting is applied to the on-air signal to prevent it from exceeding the levels that the transmitter can handle. It can also be applied to vocals or mixes –but should be approached with care because of its potentially drastic effects.

Gating
Gating is the process where a device reduces the signal (usually cutting it off entirely) when the incoming level falls below a certain point. This is sometimes used on vocal tracks in order to eliminate background noise when the speaker is not speaking.

Ducking
Ducking is the process whereby a given signal is turned down according to the level of a different signal. Typically used in some broadcast studios to automatically bring down the level of the music when the announcer speaks.

Expanding
Expansion is related to gating where an incoming signal is reduced when it falls below a certain level, but in this case it is not turned all the way off. This can be used as an alternative to gating where a gate would sound un-natural. It's called expanding because it in fact increases the dynamic range by increasing the difference between the loudest and softest signal allowing the unwanted signal to fall below the threshold of hearing.

We'll spend most of our time looking at compression and limiting since these are the processes used most commonly in radio production. Compression in particular is one of the fundamental tools used production and is one that you must become completely comfortable with using in order to achieve a polished and 'fat' sound.

Compression

Compression is an incredibly useful tool for a number of reasons.

- It makes your levels more consistent making things easier to mix.
- It helps you get your material sounding as loud as possible.
- It can help 'seating' vocals in a mix.
- It can help 'glue' a mix together.
- It can make a sound, or your mix, seem more polished, tight and 'in your face'.

How it works
Compression is the process of applying gain reduction (turning the signal down) according to how loud the incoming signal is.

A compressor will start modifying the signal only when the signal's level exceeds the **threshold** you've set – until the threshold level is reached, the signal passes through unmodified.

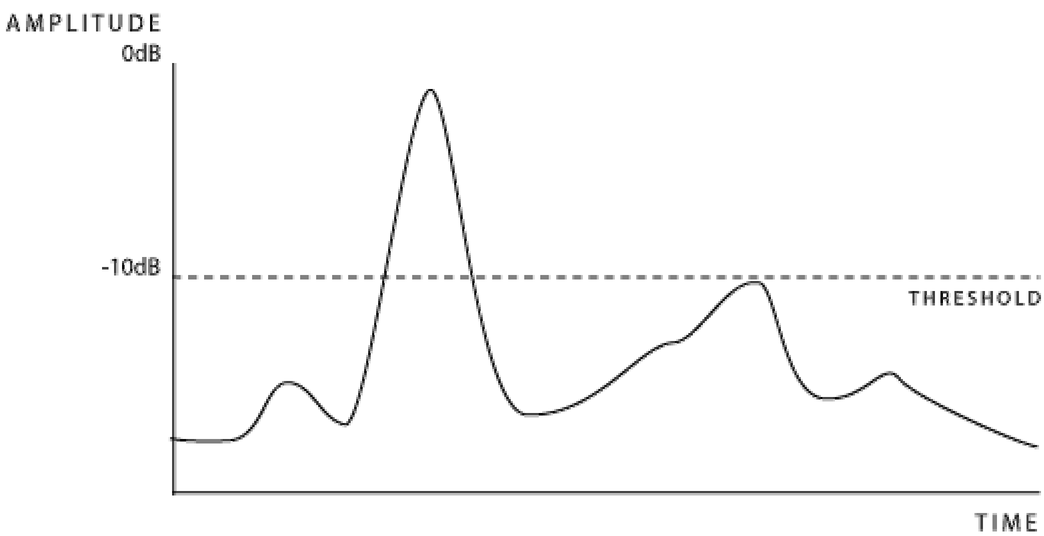
Once the threshold level has been exceeded the amount of gain reduction applied depends on where you've set the **ratio** control. This determines the ratio of incoming signal to outgoing signal (above the threshold). So if you've set a ratio of 4:1 and the signal is 8dB above the threshold, the output level will be reduced to 2dB above the threshold level.

The **attack** and **release** controls determine how fast this gain reduction is applied to the signal and then how fast it is allowed to return to normal again after the signal falls below the threshold.

Often, when compressing, the final signal will not be as high as the incoming signal since it's now being 'turned down' from time to time so there is usually also a **gain** control allowing you to boost the now compressed signal to bring it back up to where you want it.

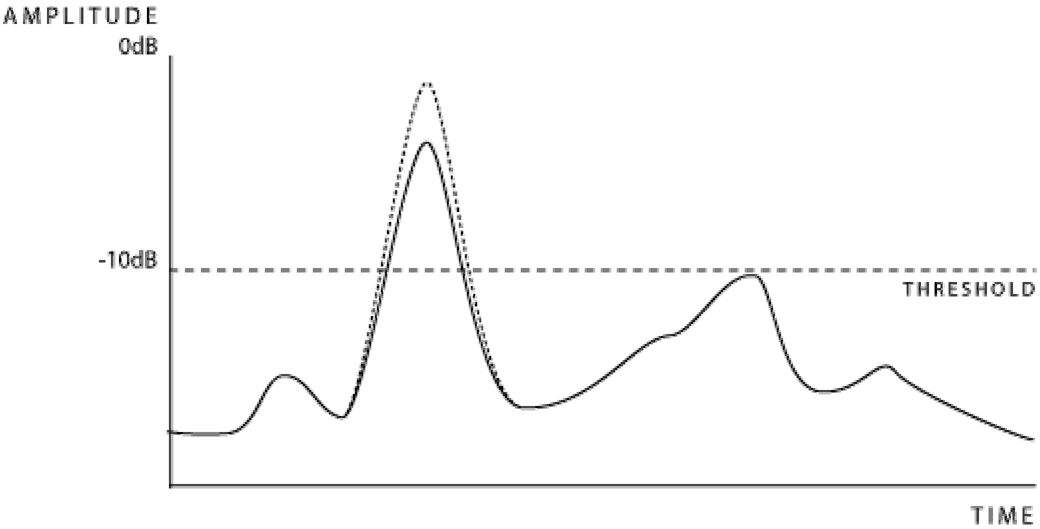
So...

Threshold
The level at which the compressor begins to apply gain reduction. The lower you set the threshold, the more of the signal will be compressed. This is expressed in dB. So a threshold of –10dB will compress everything above –10dB.

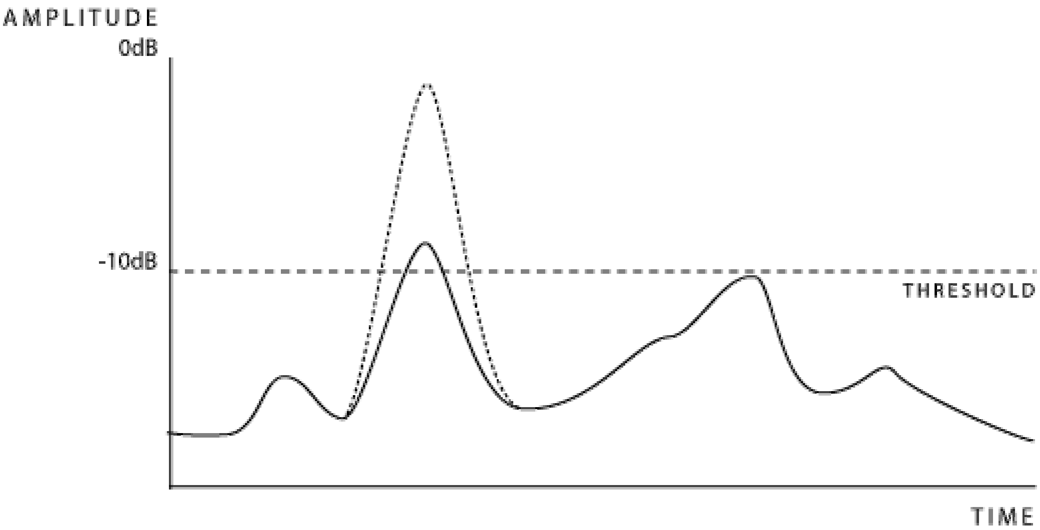


ORIGINAL SIGNAL – *NO COMPRESSION*

Ratio
The ratio of incoming to outgoing signal level (above the threshold). For example a ratio of 2:1 means that if the incoming signal is 10dB above the threshold – the level will be reduced to only 5dB above the threshold.



2:1 COMPRESSION APPLIED



10:1 COMPRESSION APPLIED

For example:
- 1:1 - no change in signal level since for every 1 dB of level coming in, 1 dB of level goes out.
- 2:1 - a signal 2dB above than the threshold will be reduced to a level 1dB louder than the threshold
- 8:1 - a signal 8dB above than the threshold will be reduced to a level 1dB louder than the threshold.
- 20:1 - a signal 20dB above than the threshold will be reduced to a level 1dB louder than the threshold.

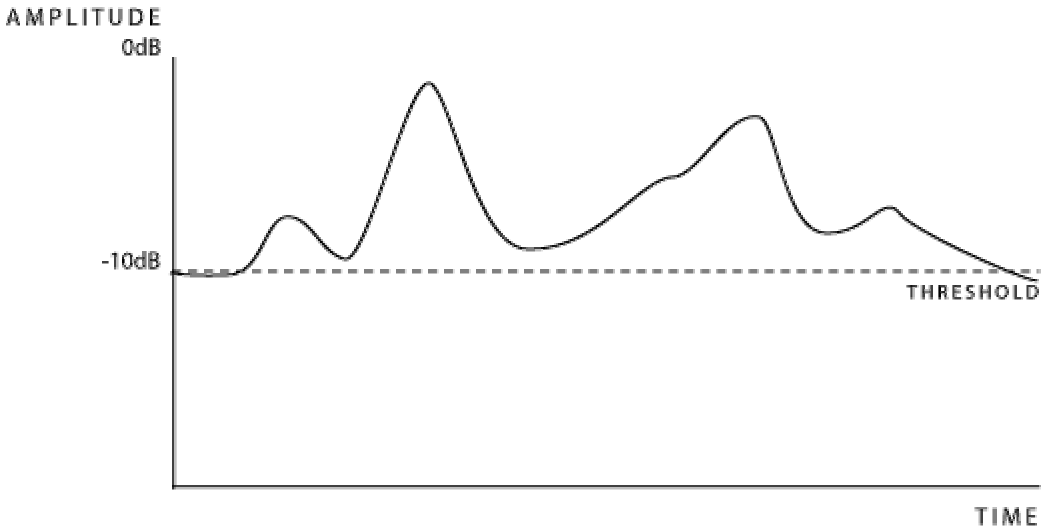
Cool Trick

If all these ratio numbers tie your brain in knots – try reversing the order of the numbers – now you get a fraction – which is the level the original signal will been reduced by.
So...
2:1 becomes ½ - or half as loud (above threshold).
4:1 becomes ¼ – or one quarter as loud (above threshold).

Attack
Determines how fast the gain reduction is applied (the volume turned down). If set to slow, some peaks will slip past the compressor before being clamped down on. This can sound more natural but means your signal is not quite as controlled. It can also mean that words with a strong attack – like beginning with a 'p' or a 't' can sometimes jump out and seem louder than the rest of the vocal. If set to fast, the compressor clamps down as soon as it can – this means your levels are very controlled – but the attack of notes are now not as dramatic, and sometimes can result in an 'un-natural' sound.

Release
Determines how quickly the gain reduction is removed after the signal falls below the threshold again. Set to quickly means that the compressor more accurately follows the dynamics of the signal, but can also result in what is called 'pumping' – obvious changes in background noise as a result of fast increases and decreases in volume by the compressor. Slower release times sound more natural, but may not track the signal as accurately and therefore some quiet sounds may become lost in the mix.

Gain
Allows the overall level of the compressors output to be adjusted. After gain reduction is applied the signal (although now more even in level) will usually be lower than originally. Gain is used to compensate for this. Once you've compressed your signal and are happy with the sound, turn the compressor gain up until it's looking healthy on the channel meter.



COMPRESSION AND GAIN APPLIED

Additional controls sometimes found on compressors
Some compressors have quite different design philosophies and controls. Some of the older style compressors don't have a ratio control at all and just compress more as you feed more signal into them, and some fancy ones have a whole host of esoteric knobs and switches. While you may always stumble across some bits of gear with controls you have never seen before, here is a brief description of some of the more common things you'll find on the more up market compressors. And don't worry too much - all compressors work on the same basic principles and as long as you let your ears guide you – you can't go wrong.

Hard Knee/Soft Knee
The Knee relates to how hard the compression is applied after the signal reaches the threshold – Soft knee begins to apply gain reduction gradually as the signal approaches the threshold and increases gain reduction as the threshold is exceeded. Hard Knee applies gain reduction as soon as the threshold is reached. Often using a softer knee (Higher values if you have a moving knee control) can mean the compression is less obvious and sounds more natural – particularly when compressing heavily.

Sidechain
A sidechain allows the compressor to be triggered by a different signal than the one your compressing. This principle is also the basis of ducking.

Auto Level
This is kind of an idiots setting which people use when they can't be bothered to actually tweak the settings out themselves. These settings often get things wildly wrong and should be avoided at all costs.

How to Compress a Vocal

Before applying compression, Eq your signal. If you are going to do any tone altering you want the compressor working primarily on those frequencies you have bought to the front, not the original signal.
Apply a mono compressor to the signal. Now, with the audio playing, set the ratio to about 4:1 and the attack and release to nominal positions (say; attack 10ms, release 50 ms). Now slowly move the threshold down until you see gain reduction occurring on the meter. Keep moving down until you see about 10dB of Gain Reduction occurring on the peaks of your signal.
The signal should now be significantly quieter than it was so grab the gain control on the compressor and move it up until the output meter peaks as high as it did before compression. Once this is done, hit bypass and compare the signal. Notice how much 'louder' the compressed signal seems? Notice also how now there is considerably less difference between the loudest and softest signal on the output meters than the input meters. Just what the doctor ordered.
You can also try just listening to the qualitative difference of the compressed signal and the plain signal by adjusting the compressor output gain until the compressed and bypassed signals sound as loud as each other. Now listen to the nature of the sound - the compressed signal should sound more punchy, tighter, and controlled.
If there are occasional bits of signal, particularly on the front of words that seem to be escaping the compressor and leaping out a bit, reduce the attack time.
If the background noise is noticeably 'pumping' up and down – increase the release time, or back off the threshold somewhat. For an even more controlled signal, increase the ratio. Observe the way the incoming signal moves much more than your compressed signal.

Have a play now and listen to the qualitative difference compression makes to a voice. When is it too much and sounds un-natural? When does it distort? What subjective effect does playing with the attack time have? What about the release time? What does changing the ratio do. Your ears should be the guide here. Do what you have to do to make it sound good. Try listening to the vocal against backing music, notice you can get away with harder compression settings when there's music playing than you could otherwise.

Once you've finished playing around, and reckon you've got a nice controlled and punchy vocal sound, set the gain so that the compressor output is nice and hot and almost goes into the red (but never quite does).

How to Compress a Mix

Mix compression can be a dangerous thing when used badly since it is easy at this point to completely sabotage your beautifully and painstakingly produced audio in one fell swoop. So use with care – and always, always, use your ears as a guide to what sounds right.

Often mix compression is used to fix things that aren't otherwise working – and if you're a real master – you won't have to use it all coz all your levels are doing the right things anyway. Sometimes you won't want to compress the mix coz it's not appropriate. Real audiophile stuff, or stuff with a lot of intentional dynamics perhaps won't want mix compression at all. Use your judgment as to how 'in your face' you want to go.

The idea with mix compression is to do two things:

- Firstly, to reduce the dynamic range of your piece so it can be nice and LOUD on air. The more you can get you levels sitting near the top of the meters, the better. Of course, there is a downside to this – it's very easy too compress to much, leaving your audio sounding squashed, flat and lifeless (sounds like a shampoo add doesn't it?). Again – use your ears – your audio wants to sound exciting and punchy. Do what it takes to make it so.
- Secondly, make the loudest thing in the mix – which should be the vocal - trigger gain reduction in the compressor which in turn moves everything else in the mix down a little and out of the way of the vocal. This helps the vocal sit nicely in the mix and will improve intelligibility. This will also make the vocal seem more part of the music and the whole mix sit together since everything is now moving around and 'breathing' or 'pumping' in time with the vocal. This is a subtle effect, but an invaluable one, often referred to by engineers as 'mix glue' and can really help make your spot sound polished and tight.

I recommend using fairly gentle mix compression settings. Try setting the threshold at about –10dB or so with a ratio of 2:1 or so – or less! This should result in your vocal triggering about 6dB of gain reduction at the most – more is probably going to be too much – but hey – maybe that's what's needed in that case – there are no rules here. Only what your ears dictate.

Limiting

Limiting (also called brick-wall limiting) is a more radical form of compression where the incoming signal is restricted, point blank, from ever exceeding the threshold. This can be useful when used carefully to iron out dynamics that a compressor just can't seem to grab, or getting brutal and just squashing the hell out of things. Using limiting on vocals after compression can give you a whole extra boost in volume and punch, but be very careful – too much limiting can make voices sound very 'squashed', tinny and painful to listen to. Similarly for mix limiting – you can get a lot of extra level out of your piece, but again, unless you are very careful, your spot can end up sounding like sandpaper.

How to Apply Limiting to a Voice

Read the paragraph above again before applying limiting – do you really want to go there? If so...

Insert a limiter after all the other processing on the voice, EQ, Compression etc... (but before reverb if you are using any).

Now gently move the threshold down. You should almost immediately get a significant increase in volume. Stop there. Quite severe limiting can already be occurring before you see any movement on the meters at all. If you are seeing any more than a few dB of gain reduction on peaks you are crushing the hell out of your vocal and it is probably starting to sound really nasty. Play around – see how far you can go – but always come back to more a natural sound for your mix.

Cool Trick

In Pro-tools LE and up - If you have inserts on insert points 3 and 4 and then decide you want to add an additional insert to the end of the chain - fear not! You don't have to reselect new inserts in their new positions and recreate all your settings. Just click on an insert and drag, and you can move the entire insert, with all it's settings around to where you want it. Including other tracks!

How to Apply Limiting to a Mix

Insert a limiter onto the mix bus (after compression, if applied).

Now gently move the threshold down. You should almost immediately get a significant increase in volume. Stop there. Quite severe limiting can already be occurring before you see any movement on the meters at all. If you are seeing any more than a few dB of Gain reduction on peaks you are crushing the hell out of your mix and is probably sounding to sound really nasty. Play around – see how far you can go – but always come back to more a natural sound for your mix.

Dynamics Philosophies

In the world of audio – you can only ever go so loud. There will always only be a given dynamic range you can use. The dynamic range of any given medium is determined by the difference between the noise-floor and the maximum operating level of the medium. Below is a chart showing the dynamic range of various different mediums.

Audio Device/Application	Dynamic Range
AM Radio	48 dB
Analog Broadcast TV	60 dB
FM Radio	70 dB
Analog Cassette Player	73 dB
Video Camcorder	75 dB
ADI SoundPort Codecs	80 dB
16-bit Audio Converters	90 to 95 dB
Digital Broadcast TV	85 dB
Mini-Disk Player	90 dB
CD Player	92 to 96 dB
18-bit Audio Converters	104 dB
Digital Audio Tape (DAT)	110 dB
20-bit Audio Converters	110 dB
24-bit Audio Converters	110 to 120 dB
Analog Microphone	120 dB

Now a symphony orchestra, on a good night, can exhibit as much as 110 dB of dynamic range. So how do we squeeze this onto a CD, or FM Radio, or even AM Radio? Compression of course. Compression is the classic tool for squeezing more information into a given dynamic range.

Now think about this: In order to really punch through on radio we want our signal to be as close to the top of our dynamic range as we can get it. If in a car, for example, we only hear stuff in the top 8dB of the dynamic range of the radio – we want to get all the information we can into this zone. How do we do this? You guessed it – compression.

Ever noticed how the adds on TV always seem a lot louder than the program material? In actual fact they're not louder at all. Whereas the audio level of 'Neighbours' might wander up a bit as the dramatic music kicks in when Harold realises he's lost his glasses (the drama, the drama) but the audio level will probably spend most of it's time near the middle of the available dynamic range giving itself headroom – or the ability to get louder when it needs to. Advertisements on the other hand – want to get their message across and 'in yer face' as much as possible so they generally compress the hell out of their audio and stick the whole lot as close to the maximum level their allowed. It's not technically louder, just more compressed, and closer to the top of the dynamic range. Geddit?

So compression is a useful tool in that, by reducing dynamics, it allows us to get more of a given signal near the top of the dynamic range. For example, if a vocal recording had a few tiny peaks at –2dB, but most of the useful signal was at –16dB, increasing the level of the uncompressed signal to the maximum allowed by the digital medium of 0dB would still leave most of the signal languishing at –14dB. Were we to compress this though so that the Peaks were now only 6dB above the body of the signal, we could now move this up so that most of the information was at –6dB. Useful.

Another thing to consider when mixing for radio is that there is already quite heavy compression and limiting applied at the transmitter to all audio broadcast. This means that whenever you compress something, it is going to be squashed even further by the transmission processing. And as my personal audio guru, Michael Stavrou points out – compression ratios are **cumulative** and **MULTIPLY** with each other. So if your vocal is compressed at a ratio of 4:1 and then your mix is compressed by 2:1, and then the on-air compression is set to 10:1 the total ratio of compression applied to your mix could be as high 80:1. Frightening huh?

So there is definitely a dark side to all this. Many people are becoming increasingly worried about the trend towards music and mediums (such as commercial FM radio) which lack any dynamics whatsoever. Pop producers have for years now been trying to make their records louder and louder – meaning more and more compressed and limited and squashed up near the top of the medium's allowances. Listen to a Ricky Martin track. There's no bit that isn't turned right up full. Record a piece of music like this onto a digital recorder and then also record a long passage of Classical music. See the difference in dynamic range. In our rush to go louder and louder – we may be slowly losing sight of one of the most dramatic and important aspects of music and sound – that of dynamic movement. The drama of a crescendo – the chorus crashing in after an intimate, breathy verse. Think about this perhaps. Squash the hell out of stuff if it needs it – but don't do it out of reflex. Allow room for dynamics in your work.

Vocal Production

Recording and producing vocal takes is an art in itself.

Don't be afraid to go for another take. In order to get a good vocal take for a 30 second spot I will sometimes record up to 10 minutes of vocal takes. It has to be RIGHT! On the other hand – some people just whack it down and rarely ever go for more than one or two takes. Your approach will depend on:

- Time – how much time do you have to complete the spot.
- How experienced and 'on to it' the talent is.
- How important it is to be perfect.
- How much you care.

I sincerely hope it is the first two points and not the last two that determine how much time you spend on a session or spot.

There are as many different philosophies and techniques for producing a vocal session as there are producers but here are some tricks, techniques and philosophies I find useful:

Before The Talent Steps In The Room

- Make sure the copy works. Read it and check dates, pronunciations, etc... to make sure it's gonna work when read. Make sure everything is typed as it will be read except numbers.
- Sort out your backing music and make sure it fits the vibe of the piece. Play some to the talent when they arrive so they get an idea of what energy level they need to produce.
- Make sure the copy is typed out and easy to read – if you're using Word, use a really easy to read font, and BIG. (I use 'courier new' 18 or 20 point and bold). If it has to be handwritten – get the talent to write it out.
- It's up to you that the spot works – if you think something is not right with the talent, copy, idea of the spot etc... fix it or get it fixed. As a producer you are solely responsible for the final outcome. If you know there's a problem and you go with it – it'll come back to haunt you. Exceptions to this are of course, time restrictions, and difficult clients who insist on stupid copy – in both cases – good luck.
- Pick your voice – every voice will be good for different things – think, young, old, sincere, sarcastic, emphatic, lazy, back, in your face, boy, girl, etc... What does the spot, and the music need.
- Get the talent a copy of the copy (!) to read before hand – preferably a few minutes before they step into the booth to record.

Cool Trick

Sometimes it can take a while to nail a particularly tricky paragraph or combination of words. Proffesional voice-over artists use quite involved warm up techniques to 'limber up' their voices boxes and their tongue to get their voices ready to record.

A quick and easy way of doing this is – get your talent to chew some gum 10 minutes before the session. This exercises the jaw and the tongue and get's them warmed up and nice and flexible for the session. Keep some Green Extra in a draw for these occasions.

Also – talent should avoid drinking milk, or milky drinks before a session. Milk coats the vocal chords and can make a voice sound a little weak and lacking in 'bite'.

In The Vocal Session

- Level check by getting the talent to read the first paragraph of the copy – if they're just going 'test, one, two' people will often then go much louder once they're actually going for takes. Otherwise ask them to count to 10. Don't give them time to think or be bored or get embarrassed.
- In all recording sessions - the talent should never have to wait for you unless you checking for takes at the end of the process. Keep ahead of the game. The talent comes first – if, in any session, the talent is waiting for you – you've messed up.
- As the talent is going for a take, close your eyes and imagine the take as actually being in place – mixed and eq'ed and sitting with the music – while the take is being recorded. If something sounds wrong here 'in your minds ear' – it is. Find out what it is, fix it, and go again.
- Be encouraging and relaxed. In order to get the best out of your talent they should be feeling relaxed and confidant. Don't belittle their efforts. Joke around, depreciate yourself – make it a fun process – and tell them when they've just nailed a perfect take. Inexperienced talent often take time to settle into the routine of doing takes and may be embarrassed by their stuff ups– explain to them that this is totally normal and you are happy to spend as much time as it takes to get 'the take'.
- Use positive terms when discussing what they could have done better – 'that was good, but I reckon we need to emphasise this bit...'. Never, ever tell them it was a really bad take unless you're very sure they will not get embarrassed about it.
- You should be able to instantly recognise 'The Take'. You should also be prepared to be able to get by if you never get it.
- Always record the first take – some people just give great first takes – if there's a little stumble here or there perhaps you will be able to patch this up, but go with the majority of this take.
- After the first take work in sections – a paragraph or one or two lines at a time is best.
- If a certain word or part of a phrase is proving problematic – change it. As long it's a minor change of a word or two and doesn't change the meaning or the information in the copy – you should be able to do this. Ask what language the talent would use to say the phrase – and go with that.
- Never start a vocal take from the word you want. If you're just after a certain phrase or word, have the talent read from a few words, or a line before the phrase so that they're breathing and intonation is in sync with the whole phrase. You can often get away with not doing this if you're recording a whole paragraph because the paragraph will probably want to sound like it's a new thing anyway, but definitely do this for individual words or phrases in the middle of a paragraph.
- Try not to interrupt a take – even if it's clear to you that it's not gonna work for one reason or another. Let them finish the take – allow them to stay in 'the zone'.
- If you're having problems getting the takes to sound convincing, ask the talent to imagine a good friend right in front of them. That's who they should be addressing – their mate – right there in the studio.
- Most people exhibit kind of an arc of performance – they get better over a few reads of each bit as they get used to the copy and the inflections and tones you want from them, and then at some point, they peak, and get tired. This can happen over a few minutes so try and get the bits you need down at the peak of their performance ability.
- Often it will take a few reads for them to get the inflections and so on right – if this is the case, they may not need to be at full energy for the process of getting to this point. Once they've nailed the technicalities, say – 'cool – now give me that with more energy'.
- If they get tired but you need to keep working – stop for a minute. Don't let them leave the booth, that takes them right out of the zone, but get them to take their headphones off, shake their arms and legs, jump up and down on the spot for 10 seconds or so – then straight back into it – that's often when you'll get the best takes. Slightly freshened up – but still in 'the zone'.
- Get to know how your talent works – perhaps they are first take wonders – maybe they like a lot of level in their headphones – perhaps they like working without headphones – there will often be variables of their performance that they are not even aware of. Observe and learn these.

- Get them to wave their hands around – if you want emphatic and excited they should be moving!
- Think and learn about inflection, tone and vocal register. A question goes up at the end. The end of a statement goes down. Think about the different characteristics of voices people use when they speak – are they excited, trying to convince you of something, or departing information? Are they your friend or authoritative?
- For FBi productions – avoid '*the radio voice*' at all costs – you know what it is – NEVER GO THERE! FBi is informal and friendly – not trying to sell you a set of steak knives. I have found that all people who have ever done any training as voice over talent – are totally unusable for FBi Voices. Fresh and inexperienced is sooooo much better than that tired old radio hype. But it can of course be a lot harder to work with.
- Develop a language to describe what you want and use it, for example:
    - "Try to sound like you're trying to be cool, but underneath you're really excited."
    - "This word should have all the warmth and affection you can muster – think about things you love when you say it."
    - "The energy level of that was good, but some of the words were a little rushed, pace yourself a bit more and try and stay in the upper register of your voice."
    - "This line here is explaining that bit above – make it sound like you're referring to something you've already said."
    - "This is an ending – come down on the end of this word."

- Don't be afraid to pull out. Sometimes it just won't work – either the talent isn't right for the read, they're having a bad day, the copy is wrong, whatever. Do everything you can to make it work – but recognise: sometimes you'll just have to go again. Thank the talent – ideally don't even tell them you're not going to use their takes, and then fix the issue. Re-write the copy, pick a different voice, whatever – and go again.

If you can, get the whole vocal session in one waveform. Then as soon as you hit stop and the talent leaves the room, gain it up if you need to, and then - NAME IT!

Vox Pops

The term Vox Pops comes from the Latin term 'Vox Populi' meaning 'voice of the people'. In Radio parlance it means recording people on the street, or in an outside environment (club, venue whatever). Recording vox pop's is an art in itself and can give great results when done right.

Often you will record 20 vox pops and get two that are good – be prepared to recognise when some-one is unlikely to give you anything good, and then really milk people who are enthusiastic and articulate, or have some kind of personality which you think will work well on air.

Here are a few tips:

- Dynamic Microphones are good for Vox Pops since they are:
    - More physically robust.
    - Less sensitive so you don't as much background noise .
    - Are often designed specifically for recording vocals.
    - Don't require there own power source.
- Think about what you want – often you can use tiny fragments of stuff that you can contextualise in the piece – 'yeah it's great', 'I love it' etc… These can come to mean anything when cut into your piece so develop questions that result in the answers and vibe you want - they may not be specifically the question that you will be presenting in your piece at the end of process. So – 'Are you enjoying the music tonight?' 'What do you like about this music?' Will often generate far better stuff than 'Do you like Sunsets on FBi?'

- Be silent while the person is talking! In normal conversation if some-one else is talking, people usually interject lots of little half words like 'yeah', 'uh-huh', 'right' etc… to indicate that they're listening. This is often completely unconscious. Learn not to do this when Vox Popping – you want just the person speaking.
- You don't have to get a good recording of the question each time – if you are in the same recording environment (background noise etc…) you can often record a few good takes of the question on it's own (if you intend to use this at all) and then cut it in later. It's more important to have the mic ready to catch whatever the 'talent' says as soon as they start, so have the mic pointing at them already when you ask the question, or at least by the time your coming to the end of your question.
- People respond best to enthusiasm and cheerfulness, so get some strong coffee in ya, and be bouncy and engaging – this will result in people more likely to want to talk to you and better responses when they do.
- Get in a flow – try and get in 'the Zone' and stay there. By hitting people up quickly one after the other – this helps you stay in the zone and elicit better responses from people.
- Work out your spiel so you can just rattle it off: 'Hi, I'm from Radio Station FBi – do you mind If I ask you a few questions?' And then have to or three questions ready to go.
- Develop your questions and modify them as you find them working or not working , but if you follow a pattern, this will make editing easier later.
- Vox Pop's are a great way to get energy and vibe happening – often you can use tiny little scraps of stuff that you may not have realized at the time were so great. Just some-one going – 'hell yeah' or 'Are you serious' can be great! Go for vibe!

Cool Trick

Wild Sound!

Whenever you're doing recordings in any environment where there is background noise; be it dialog for a film, an interview with a musician, vox pop's, whatever…

ALWAYS RECORD SOME WILDSOUND!!!!

This means record just some of the natural ambience of the environment without anyone talking. Record at least a minute or so.

This will become incredibly useful later to fix holes when your editing – if there's a weird pause or the levels change because you've moved or anything – you can now hide this, or cover it up by laying some wild sound alongside. You can also use it underneath a vocal you record later in a studio to make it sound like you were there, etc…

Invaluable!

Building A Spot

Putting together a nice tight bed for your spot can be a surprisingly tricky task. Sometimes you will have extremely specific instructions on what music to use from a client – right down to the second. Other times, you will be given copy and have to source the music for it yourself. Here are some ideas to help with this task.

Guidelines For Bed Selection

- Always try to go with Sydney music if you can, and if not, then at least Australian.
- Choose tracks that complement the intent of the piece, if it's a spot about a band competition, using techno may be quite right, etc...
- The music you use should always be polished and punchy and make you feel good.
- If it's an arts thing – perhaps go with something more meditative and experimental. But maybe not – arts can be funky as hell. But maybe not beer drinking songs...
- Try to think if there are any specific songs or bands that have some kind of relationship or relevance to the spot – do you know of a song with a lyric that perfectly matches a line of the copy? Or that is thematically linked – can you work this into the copy? If you can – go for magic.
- If the spot is an FBi Station thing – try to cover both the electronic and rock idioms. Ideally find pieces that are kind of halfway between both – like beats with guitars, or grab one piece of electronica and one piece of rock.
- I tend to use about two bits of music for a 30-40 second spot. Sometimes just one, sometimes three. I find it helps keep the spot exciting if you build tension with one track and then resolve it with another. Or similar. Just helps keep things moving along.
- If the spot is for a music artist or gig – get the best bits – this means the loudest chorus (usually the last) off the biggest singles off the latest album. Try and get a couple of really identifiable, catchy hooks.
- You will need to find bits of audio without vocals.

Golden Rule

Never, ever put a vocal on top of something that already has a vocal!

Just don't.
It doesn't work.
Mmmkay?

- This means that you will be looking for:
  - Intro's: A good long intro can be great bed material – particularly if there's a nice big dramatic drum fill or the like towards the end of the section that you can use to launch you into the next piece of music in the spot.
  - Breakdowns: fast forward to the middle of the piece of music – is there an instrumental verse? Or a solo of some sort? (be careful here – guitar solos can clash with voiceovers almost as much as other voices do.)
  - Outro's: Great for the end of a spot. If you've found a piece of music you want to use as a bed, always go to the end of the song and see if there's an ending you can use. If there's no vocal hanging over, the best ending to a spot is always a nice tight band ending with cymbals and everything.
  - Random bits of noise! Once you've found your piece of music, grab things like– the two seconds of guitar feedback at the beginning of the track, a really wacky noise from the middle of the song when nothing else is playing, the huge drum fill going into the last chorus. These can be used to punctuate different bits of the copy or help transitions from one section of the copy to the next – try and grab bits that are memorable and exciting.

  - Electronic and dance music often lends itself more to beds since it is often instrumental, and easy to loop and edit with. Again – look for sections that end in an obvious crescendo, fill or something that you can use to launch into the next section.

Once you've got your bits of beds to play with, you should record your voice over. Remember to play the talent the music you intend to use so they can get any idea of the energy level of the spot. Once this is recorded, and you've compiled the best takes into one track of the entire copy for the piece and have the voice all ready to go it's time to construct your spot. Here's where it gets fun!

Guidelines For Spot Assembly

- Keep it punchy – there should never be a second without something happening – either the voice telling you something, a really attention grabbing piece of audio, or the chorus or vocal from a song. This is ALL IMPORTANT!

Cool Trick

Time compress your vocal!

This can be useful for a number of reasons:

- **It makes your vocal a little tighter and therefore feel a little more high energy.**
- **If you have a lot of copy (and you often will) it can help squeeze it all in.**
- **If you have a section where you are trying to get the vocal to sit in with a certain section of music, like 8 bars, a bit of time-compression can get you an extra half second here or there.**

Make sure the effect is never noticeable, for most reads a maximum change of 1: .96 is the absolute maximum you can apply before things start sounding rushed and un-natural – but experiment here – see what works. Also make sure your sound quality/rhythm fader in the Audiosuite plug-in is set all the way to sound quality.

AND USE HANDLES – see page 30.

- If the spot is to promoting a music artist, or something musical – allow space for the artists music to do it's work. So in a 30 second spot have at least one, if not two big choruses, or bits of them in between sections of the copy. If you can find a bit of a verse that doesn't have vocals, that then crashes into a big hooky sung chorus this is ideal coz you can put your vocal over the first bit, and then let the chorus swing up and do it's thing. Otherwise you may have to fake it by using an intro or other instrumental bit and then editing the chorus onto that. The drum fills you pulled out earlier may be useful to help transitions like these.
- You only need one line of a chorus though. Particularly if there's a lot of copy – all you really need is enough for people to recognise the song, get caught up in the energy and then bang – edit into the next section of the bed and the next piece of copy.
- When a piece of music crashes in, or on transitions, allow the music to establish itself for a little bit before the vocal comes in. The first word punching in right on the 'two' is often good. Count it out and try and edit the vocal so that it sits 'in time' with the music.
- Also allow transitions (from one piece of music to another) to breathe. Unless you are trying to hide something, don't put vocals over transitions, use them as a bit of a breathing space from the copy and a place where something interesting happens before launching into the next bit.

- Punctuate the copy with bits of noise. Once the basic structure of your spot is in place, feel free to scatter some incidental bits of noise throughout to make it more interesting. If it's a live gig perhaps have a track of crowd roar just quietly in the mix that surges out at a transition or at the end... Or at the end of a particularly dramatic line of copy drop a piece of guitar feedback – run backwards. Or a screamed 'yeah baby' from the intro of another track. God is in the details. Try Eq-ing a particular piece of the voiceover into the radio voice – or into a wacky effect (careful with this though – you can very quickly end up sounding like Nova with this one...). You get the idea though – let your creative juices flow.
- Fix it with a whoosh! Sometime you will have to do a transition from an 80 bpm acoustic thing into some 130 BPM high energy techno. Trust me – there will be a time when you have to do just this and MAKE IT WORK! So sometimes, just sometimes – you have to fix it with a whoosh. Whooshes are seen by some to be as much of a copout as I see fades. But I am partial to the occasional well placed whoosh and/or bang. It's all about context really. And it's about the type of whoosh – the best whoosh of all will be one that comes from the song itself that you're working on, or perhaps something created from the voice you're working with, something that is relevant tonally to your spot. And remember a whoosh can be anything – and a backwards whoosh is a bang and vice versa. If you have to - use whooshes, but do it with class, or get sneaky. *Swish ---- BANG!!!!!!!!!!!!!*
- If you have to edit into a section with no cymbal or sound to punctuate the beginning of that section – stick a bang in. Keep a little collection of crash cymbals and the like handy for these kind of applications, and they don't have to be big, just sit them back in the mix, so that they blend into the bed.
- Starting the spot off with a dry vocal – even if it's just a word or two - can often be a good thing coz it allows the presenter to 'marry' the spot with the previous piece of audio. Otherwise - come in with a bang! Avoid fade-ins or weak, soft sounding intros.

Cool Trick

Use 'Handles'!

Handles are areas of an audio region outside of the bit that you actually want to use. These allow flexibility in that if you later decide you want to use a bit more of an region than you first intended - it's there. This particularly applies to when generating regions using Audio-suite. If you gain something up, and then later decide you just want another half second after the word to marry that piece against something else – you're stuffed and will have to go find the original bit and do the operation again, move it into place, etc...

Unless you create some handles for yourself.

When about to apply gain, drag the left and right sides of the region out a little from what you actually intend using. 2 seconds if it's a short bit of vocal, 5 bars if it's a piece of music, or 10 seconds if it's an ambient recording. Perform the Audio-Suite process, and then resize to what you need right now – trust me – get into the habit of doing this and you will be thankful later.

- Go out with a bang! Try and find a nice punchy dramatic end to the spot. Fades are for wimps.
- Edit on the one. Try to edit together music with a real sense of rhythm throughout the spot. If you can sit and count 1,2,3,4 throughout the whole piece and end on the 1 – fantastic. If you can do that and then deliberately have it all fall apart at a certain point for dramatic effect – even better.
- Think about the spot like a song – you should get the listener involved, create tension, and then resolve it.
- Think punchy – if there's ever a bit of the spot where all of a sudden the energy drops and it all sounds a bit weird or weak – fix it. It's gotta be 30 seconds of magic.

Cool Trick

Hiding Tracks!

In Pro-tools, a convenient way to temporarily silence a piece of audio without having to go mess around in the mix page is to 'hide' a region.

Do this by simply highlighting the region you want to hide with the grabber tool, and then hit APPLE+M on the keyboard. That region will now go kind of faded out on screen and will no longer play. All mix variables etc... remain untouched.

To 'un-hide' the region, just select it again, and hit APPLE+M again.

Like most other things in audio – there are as many different ways of doing things as there are producers. And very, very few golden rules (the voice on voice thing above is a notable exception). So experiment, try out different things – check out how commercial stations do it. Now check out how community stations do it. What elements do you like of both? What elements of both do you dislike. And most of all...

Listen and think!

Pro-tools Automated Mixing

Mixing starts to becomes a real pleasure when you grasp and begin to use the automated mixing functions Pro-tools has to offer. I recommend reading the automation chapter in the Pro-tools manual from start to finish. But below I will outline some of the more important functions and concepts that you should get completely comfortable with.

Volume Graph in Edit Window = Fader Moves in Mix Window
Most of you will already be using the ‘volume’ display mode of the tracks in the edit screen to automate volume moves on your tracks. If you enter some volume data this way, go to the mix window and watch the faders move. The lines you see in the edit window are graphical representations of how the faders move on playback. This can also work the other way. Click on the automation state button (above the record button on the channel strip) and change this from ‘auto-read’ to ‘auto-touch’. Now, whatever movements you do with the fader while Pro-tools is playing will be recorded and played back from then on, and can be seen and edited as ‘volume data’ in the edit window when your track is set to display volume.

Use Auto-touch mode to build mixes
Once you’re into mixing your piece, I recommend setting all the faders to ‘auto-touch’. Then simply go for passes moving levels around: wiggle the faders of the music under the vocal, move the vocal up to catch a certain word that’s a bit unclear, etc... This is the way to build dynamic, exciting, ‘perfect’ mixes.

More Uses For Auto-touch Mode
‘Auto-touch’ mode applies to mute and pan too! This is the way you can record live passes of pan movements and get stuff whizzing around the stereo field. Or tap in rhythmic track mutes and record them.

If you’re in ‘auto-touch’ mode it can be problematic using the channel mutes to silence a track while you’re working on something else, since this actually writes mute data to the track. Instead get used to using the ‘hide region’ feature.

Detailed fiddling
For really fiddly work where you can’t quite get the moves with faders, set the track display in the edit window to ‘volume’ and then use the pencil tool to draw automation data into the track itself. You can get more detail if you set the track size to ‘large’ or ‘jumbo’ as you do this.

Adjusting a whole series of moves
If you have a whole series of moves (on your vocal track for example) which you want to keep, but want to adjust the whole section up or down a bit in volume, use this trick: In the edit page select volume display for the track you wish to adjust. Now use the selector tool to highlight the area you wish to modify. Now select the trimmer tool, click on the highlighted region and drag the whole area up or down the desired amount. Little numbers should appear on screen showing you how much you’re moving the data (the number in brackets after the little triangle symbol) – again select a nice big track display size for more resolution if you need it.

These are really just a few pointers to get you started, again, I recommend you read this chapter of the Pro-tools manual (included on the CD that came with these notes) from start to finish.

Mixing

Here's where the fun really starts.

Once you've sourced your audio, recorded your vocal, edited your whole piece together and it's all more or less in one place – you're now ready to mix your piece down. Mixing involves balancing all the levels, Eq'ing whatever's necessary, perhaps running some compression here and there, making the whole thing sound as polished, punchy and loud as you can and then bouncing your spot down to a stereo file ready for broadcast.

Sometimes editing and mixing can be kind of dependant on each other, and you will have already done some balancing and fader moves before you get to this point. This is fine. Often I find that I want a really punchy vocal sound to know what I'm editing around and to really be able to support every nuance of the vocal - so I will do some work on the vocal first. Sometimes certain fader moves or transitions are essential to the feel of the whole spot so you may want to spend some time on these first, but generally, first you edit, then you mix.

Mixing is an arcane art that can take years, or a lifetime, to master. There are as many different perspectives and philosophies on mixing as there are people mixing. It is a world where there is no right or wrong, and very few solid rules. Whatever works best, works best – and at times – it can be some far out stuff.

There are however some over-reaching guidelines to keep in mind as you mix your spot. While there are infinite effects and approaches to any given situation, it may be worth keeping these in mind and making sure that you take these into account.

- Loudness – you want your spot loud. Particularly if it's a normal 'run of station' spot that is going to have to compete in level with already mastered and probably heavily compressed music. Your spot should, subjectively, be just as loud as that latest track by Basement Jaxx.
- Intelligibility – your spot may be loud as hell and really exciting, but if you can't hear the phone number at the end coz the mix goes all funny and there's a guitar that drowns it all out, you may as well have stayed in bed that day. In certain situations it's O.K to just imply information and energy – but for the most part, and certainly until you're very sure of what you're doing – every word of the copy you've been given has to be crystal clear. And crystal clear on all systems – big stereos, car stereos, little transistor radios etc... Not just your headphones or those fancy studio monitors you've been mixing on.
- Clarity of intent. Whatever the spot is – it has to sound like it means it. If it's a spot advertising shoes, by the end of the piece, you want to leave the listener with the feeling that shoes are really relevant, important or cool. This is a good test to run on your work. Once you're about to bounce, sit back, close your eyes and run your piece. Imagine you're hearing it for the first time (hard I know – a good way of getting into this headspace is to listen with someone else – this somehow always focuses your mind on the spot in a new way). Now – are you convinced? Does the spot leave you feeling good about FBi and whatever message you were delivering. That's really the money shot here. If the spot leaves a smile on your dial – you're doing O.K.

Step 1 – Sort Your Vocal

Now is the time to apply all the Eq and compression knowledge you've learned previously to the most important thing in your piece – the vocal. Again here there are no hard and fast rules, your vocal might have been recorded in the studio at FBi, backstage at a concert and have lots of background noise, be part of a phone interview – whatever. The important thing is that it cuts through the mix while still sits 'in the mix' and sounds 'good' whatever that means.

For the most part you will be wanting to Eq your vocal to bring out some tops and perhaps some warmth in the low end, and then compressing it to even out the dynamics and get it nice and punchy. For that really in your face sound you may even want to limit it a bit, and perhaps – if desired – add some effects.

- In terms of signal flow, always Eq your vocal before compression since you want your compression working primarily on the frequencies you bring out with Eq.
- In terms of compression, some spots may want more than others, moody more sparse pieces may want to sound more natural and have less compression, than an 'in your face', lots going on kind of spot. Let your ears and your taste decide.
- After Eq and compression, you may find your vocal is a little too sibilant, and that the ess's are really jumping out at you. In this case insert a de-esser as the last thing in the chain of inserts and try pulling down the threshold a little – just enough to remove the harshness of the 's' sounds.
- Often you won't need a lot of bottom end in the vocal – using a filter to eq out everything below 500 Hz or so can really work and allow the vocal to be louder in the mix, but avoid making it sound harsh or brittle. Play around with this and see how much you can afford to lose before it starts to sound un-natural. Removing all the frequencies down low here is also an 'effect' kind of sound and can be good in some cases, but generally not all the time for a whole spot.
- Generally a voice doing a straight read won't need reverb, but experiment; sometimes a small amount of a really short reverb can help a vocal sound a bit fatter and more polished. But think short reverb times and use just a tiny amount. If you can hear the effect – generally that's too much.

Step 2 – Apply Mix Compression If Desired

If you are going to use mix compression, apply it earlier rather than later since this will affect the rest of your mix and how you balance your levels. For details on applying mix compression see Chapter 6.

Generally, you will want an absolute maximum of 6-9dB of gain reduction being applied.

Use the compressor gain then to set the final level of your output. Check your master fader levels as your piece is playing and set your compressor gain to be as loud as you can without ever going into the red. Generally you want quite a bit of your piece going right up to the top of the meters – but never quite over the top. The more information you can squeeze into the very top of the meters, the louder it's gonna be on air. If you've achieved this – you're doing well.

Step 3 – Balance Your Music Levels Around The Vocal

Now use the automation features to set the levels of the music in your spot. Generally, you want the music as loud as possible in order to impart energy into the piece without interfering with the intelligibility of the vocal itself. Keep in mind that you know what the vocal is saying coz you've been listening to it for half a day whereas listener will only ever been listing with half an ear, probably in a noisy environment, and not really interested unless you make them so. If you ever find yourself squinting, or leaning your head forward when a vocal is playing – that's a good sign that something is a bit difficult to understand and wants some adjustment. If in doubt, always err on the side of quieter music.

Whenever the vocal is not present in the spot – bring the music levels up – having a really dynamic spot where the music swoops in and around the vocal is generally good. But don't overdo it. You should be conscious of the music, but not the changes in level.

Use the fader automation to ride the levels of the music underneath the vocal. In little pauses between phrases 'goose' or pump' the levels of the music a little.

Step 4 Balance Everything Else

Now go and cast a final eye over all the other elements in the mix. Any additional SFX, secondary voices, or atmos etc... should now be set in with the mix. The trick here is to make it all sound like one piece of audio. Anything that jumps out and makes you think – 'hmmm what's that doing in there' is usually not quite working yet. Since this is the last stage before committing your piece to prosperity – have a final think about all your edits – does it all feel tight. Is the message totally clear? Is the beginning right? Is every word intelligible and does the music work with the vocal. Make sure it's all right now.

Panning
Check your panning. Anything that is essential to the spot making sense must be panned dead centre. You have to assume that someone listening with only one speaker will get all the important information. Having said that – use of stereo can be a great tool for creating something dramatic and attention grabbing. Incidental sounds, repeated vocals, scratches etc... Use discretion – but make 'em fly around the room. Just ensure they are adding to the interest or the message of the piece and not distracting from it. Music should pretty much always be panned hard right and hard left.

Clicks
Have a final check for any clicks at this point – Headphones are really useful for this – close your eyes, focus really hard on the sound, particularly the top end, and make sure there are no little clicks or pops in your edits.

Use Different Monitors
Now's also the time to check your piece on all the speaker systems you can. Most important are the little crappy speakers, or ghetto-blaster you've got set up for this reason. Often experienced engineers will mix almost entirely on 'little' speakers. The more you do this the better your work will be. If it sounds good in crap speakers – it will sound amazing on the bigs. If you can do a test bounce and burn it onto a CD to play in a few different systems this is ideal. But generally try and get as many different perspectives on your mix as you can.

Step 5 Final Check on Levels and Bounce

Run the whole spot with the master compression showing and watch the levels on the compressor and on the master fader itself.

You should now have a nice hot piece, where the hardest thing hitting the mix compressor is the vocal and the levels on the master fader are nice and loud.

Once you're convinced this is all looking totally cool, do your bounce. Try and name the file already now what it's going to be called when it's on the audio server.

Also have a listen the first time your piece is broadcast on air – this is the acid test really. How does it sound? Does it sound totally polished, and sit perfectly alongside the music? Is the vocal clear – are there any un-natural level jumps? Does it sound too squashed or compressed? On air compression will change the sound of your piece considerably so get used to what things sound like off-air and then on-air. Knowing this difference and being able to work with it is the key to being a good radio producer.

Now you have official permission to run around as pleased as a dog with two tails and play your spot to everybody you can drag to a stereo and generally act pleased with yourself.

But the job's not finished yet! Now go back and consult Chapter 1: 'Spot Creation Process – An Overview' and follow all the remaining steps to get your spot to air. Clear it with whoever you have to, put it onto the DRS and make sure it's named and loaded correctly, and sort out the scheduling so that the piece will actually be played. Once this is all done...

You've just made radio.

Glossary

Gain – amount of amplification applied to a signal

Pot – A rotary control, or a knob.

Hot – A high level signal.

Talent – vocalist, musician – any-one on the other side of the microphone.

Run of station – To be played at any time in normal programming.

Copy – The text for a promo or spot.

Handles – Small sections of audio on either side of the bit you are actually using.

Sibilant – S sounds too harsh or extreme, often the result of overcompression.

Goose – A little jump in level at a certain point. 'Goose it a little after the vocal'.

Root Directory – The bottom directory of any area. The root directory of your hard drive will be C:. The Root directory of a Pro-tools session is the folder named after the session.

Credits

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