Conquering EQ

By Eddie Bazil, Samplecraze.com

Contents

i. How To Use This Book

Preface – How to Use This Book

Tutorial Files

Many of the tutorials in this book have associated files that are used to explain the concepts more clearly to you. All tutorial files are included in the 'sounds' folder that came with this book and

Make sure you extract all the files correctly within a program such as Winzip (**www.winzip.com**).

Searching
The beauty of a PDF is that you can search its contents very easily from within the free Abode Acrobat reader. I recommend you install the full version (it's still free) which is available from Here

http://www.adobe.com/products/acrobat/readstep2.html

To search this Ebook, simply go to 'Edit' and select 'Search'. Alternatively, if you have a PC just press 'CTRL & F'.

1. Introduction to Equalisation

Almost every forum I visit, or every 1 in 5 emails I receive, the subject matter that is most discussed is EQ (equalisation).

What it is, how to use it and when to use, which one to use and which is the best for the given task and, most importantly, what frequencies cover certain sounds. Everyone is after a quick formula.

What most people do not realise is that EQ is one of the most powerful tools available to a producer, and mastering house, and to master this tool one needs to understand frequencies.

When the term 'EQ' (equalisation) is mentioned, people invariably think of the tone controls on their hi-fi systems. Knowing this actually gives you a head start. The fact that a sound can be made warmer or punchier already gives you an idea as to what EQ is. You know that by turning the treble knob up on your hi-fi makes the sound brighter, turning it down makes the sound less bright and more fluffy or woollen.

What you are actually doing is boosting, when you turn the knob 'up', and cutting, when you turn the knob 'down', the predetermined frequencies that are assigned to that knob.

Before we get deep into EQ, let us look back a tad at why EQ was introduced into our lives in the first place.

How and Why

This whole headache began in the broadcasting field. Blame those guys.

Actually the first ever instances of equalisation was in the communications industry. EQ was used to counteract some of the problems in telephone systems.

Tone controls were created and used to compensate technical inaccuracies in the recording chain, more notably, compensating for microphone colouration and room acoustics. EQ was used as a means of controlling the gain of a range of frequencies.

In those days, a standard, bass, middle and treble gain control were all we had. Today, we are actually spoilt for choice as there are so many parameters that we can control on any type of EQ, that it has actually become confusing.

From corrective EQ, we have come a long way, to creative EQ.
I do not want to bore you with the history of EQ or who invented what. You can trawl the net for some excellent articles on this subject. What I want to do is to dive in and get you started on this detailed and confusing subject.

However, before I get into this deeper, I need you to understand 'sound' and 'frequencies'. Those that have read my tutorials on my website http://www.samplecraze.com will know that I have covered the subject of sound in detail. For those who cannot be bothered to read those tutorials, here is the condensed version:

To understand any part of EQ you need to understand sound, what it is, how it moves, how we perceive it and why we perceive it the way we do.

Once you understand this then shaping it or manipulating it becomes so much easier.

Tagging

The method I am going to use to help you in understanding the terminology and definitions and what each part is and does is a system that has existed for centuries and works extremely well in every aspect of your life.

*It is a system that speed reading specialists, memory recall centres and even high powered executive training programs use. It is called '**Tagging**' or '**Linking**'. It is the simplest and most effective 'remembering' tool.*

You have used it since you were a child. Every time you were asked to draw a house in a field, you would draw a strong big house on green grass with a huge sun that was always yellow, red or orange with a tree and a cow. Of course some people drew the same topic in the Picasso mould or surreal a la Dali, but on the whole, the picture is almost always the same.

Why? Because we remember things that have an effect on our senses, be it touch, smell, taste, hearing or visual. The strong colour of the sun and the size of it are a great way of remembering what a sun looks like. The big house in the centre of the drawing will always stay in your mind. The cow is always alone and strongly accentuated and is always totally out of size in comparison to the house. The ground is always green grass and the sky always blue and if there is a cloud then it is always one big round cloud.

These images are strong and always stay in memory. The same technique is used in tagging. We create an image rich in as many of the senses as possible and that will always stay in our minds, far stronger than having to learn things in parrot fashion.
I have used this technique all my life and now do it unconsciously. Not only does it work but it is also fun as the tool for tagging is your imagination and nothing is stronger or stays longer in your memory than an image created out of your imagination.

Probably by now you feel that I require a great deal of help and that there are certain places for people like me, comfortable places that offer 24 hour security and in depth treatment. You are probably right but not for the reasons for this tutorial.

So, let us begin this journey.

2. Sound

Sound is the displacement of air around the source and how we perceive that displacement.

Right, what does that mean?

Think of the best and most commonly used analogy: that of dropping a stone in a pond and watching the ripples form. The ripples always move away from where the stone meets the water (source). The air displacement is the ripples created by the dropping stone. In this case we see the ripples. In the case of sound we hear the ripples (the displaced air).

How do we hear the displaced air?

Our eardrums pick up the displaced air and our brains then process the data as sound. I could go into the details about the ear muscle vibrating and the eardrum being a chamber and on and on and on…but that is not what you want to know.

The tagging image here is the **ripple**.
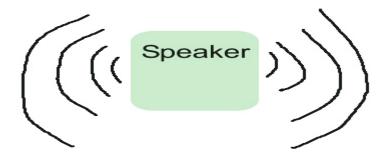
Has it ever occurred to you that when a picture is drawn of a guitar amplifier, with a guitarist playing loudly, you always see a few arced lines drawn coming out of the amplifier?

Exactly like the ripples in the pond. In fact this image is always the same for speakers that are playing music. The picture or tag here is, again, always the same with arced lines starting as small arcs growing to large arcs and away from the speaker.
Keep that image in your head and that constitutes sound, or more precisely, sound waves, like the ripples.

Fig 1 illustrates this. I have deliberately used freehand so it will make you laugh and stay in your memory.

Fig 1   SOUND WAVES

3. Components of Sound

Now lets us look at the components that make up sound.

There are **three**, and are really quite simple to understand if you apply the ripple analogy.

*The displacement of air or air pressure as is more commonly known, creates the waves in **fig 1** and is know as Sound Waves. The rate at which these waves occur is called **Frequency**.*

So our first component of sound is **Frequency**.

i. FREQUENCY

This is simply calculated at how many **cycles** (waves) occur every second. These cycles are repeated so really we only need to look at how many cycles (waves) occur in one second.

The result is measured as cycles/second and this unit of frequency is called a Hertz and the abbreviation is Hz.

You cannot get simpler than that…..how many cycles hit you in one second.
Heinrich Hertz was a dude who worked with wavelengths and frequency, so we have to thank the man and it seemed only right to name this little calculation after him. I always remember the rent-a–car agency when I think of frequencies and Hertz and it makes me smile every time so remembering that name is easy.

To give you an example of how easy this is check out the following:

If you had 50 cycles hit you in one sec then that would be a 50 Hz wave. There, simple and makes you look cool in the bar when you want to impress someone…or maybe not.
So it also follows and makes complete sense that if you had 10,000 cycles per second then that would be 10,000 Hz, but, because we don't want to have to write so many numbers every time a thousand appears we use the **k** letter to mean a thousand.

So, 10,000 Hz is now written as 10 kHz. Now you look even cooler. There is a reason we do this and it's not because we want to look deep and complicated individuals but simply because of all the work that has been carried out on our hearing range in the past.

And a range was formed, sure it varies but generally speaking, our hearing range is anywhere from 20 Hz, deep, to 20 kHz, high.

Now, let us think of that range and make life a lot easier by giving names you recognise to the frequency range.

So: bass, midrange and treble are easy to remember and if you are old enough then that's about all that used to exist on hi-fi systems back in the days of armour and jousting. Now let us give those tags a frequency range and then all becomes so much easier to understand.

Bass: 10 Hz to 200 Hz

Midrange or mid, a term you hear a lot of engineers use: 200 Hz to about 3 kHz.

Treble: 3 kHz to whatever the highest value you can hear.

It is important to mention, at this stage, my beliefs regarding frequency charts. I am not talking about the cycles chart coming up in this tutorial, but about charts depicting ranges of instrument frequencies.

I put very little importance on frequency charts for instruments. Almost every website you visit, that displays a frequency chart, you will see that they all vary in their ranges. The reason for this is quite simple. Ranges can be both inaccurate and broad. I think it is a waste of time displaying charts, because almost every student I have tutored, that has tried to use one of these charts, has still ended up requesting help on EQ.

It is far more helpful to understand frequencies and sound, than it is trying to use a chart of this type.

Once you understand the content of this E-book, you won't need a chart.

I will, of course, give examples of certain sound frequency ranges and what happens when you apply certain EQ parameters to them.

So we now know that higher frequency sounds are higher in pitch as there are more cycles per second and lower frequency sounds have fewer cycles per second. Easy.

Right now I think it is important to show you a frequency chart for all the notes on a keyboard or scale and the midi note numbers as well as this will come into play at a later date when we deal with synthesis and programming with the use of midi.

You do not need to learn this chart in parrot fashion but it is important to understand some of the frequencies that are used as, later, you will need to know these frequencies so that if you need to use EQ to shape a sound or remove or add certain frequencies, then the chart can prove to be invaluable
.
In most cases, you only need to recognise the main frequencies for certain notes. For example: C4 at 261.63 Hz is a great reference point, because then you can find, easily, C5 or C3 etc….

I cannot stress how important frequencies are for the understanding of sound and EQ. Engineers live by them as do producers and Sound Font developers.

If there is one piece of information that overrides any other in terms of importance it is the understanding of frequencies.

How often have you tried to mix your track only to be mystified by the result?

Terms like 'muddy' or 'thin' spring to mind and these are all because the mixer or producer does not have an understanding of frequencies and their effect on other frequencies in a mix.

Understand this basic concept and you will be armed with the most potent weapon.

Waveforms and frequencies go hand in hand.

Understand these two and the rest is all about using the tools.

So, let's get on with the **CHART**

4. Midi and Frequency Chart

Midi No		Note	Keyboard	Freq	
21	22	A0		27.500	29.135
23		B0		30.868	
24	25	C1		32.703	34.648
26	27	D1		36.708	38.891
28		E1		41.203	
29	30	F1		43.654	46.249
31	32	G1		48.999	51.913
33	34	A1		55.000	58.270
35		B1		61.735	
36	37	C2		65.406	69.296
38	39	D2		73.416	77.782
40		E2		82.407	
41	42	F2		87.307	92.499
43	44	G2		97.999	103.83
45	46	A2		110.00	103.83
47		B2		123.47	116.54
48	49	C3		130.81	138.59
50	51	D3		146.83	155.56
52		E3		164.81	
53	54	F3		174.61	185.00
55	56	G3		196.00	207.65
57	58	A3		220.00	233.08
59		B3		246.94	
60	61	**C4**		**261.63**	277.18
62	63	D4		293.67	311.13
64		E4		329.63	
65	66	F4		349.23	369.99
67	68	G4		392.00	415.30
69	70	**A4**		**440.00**	466.16
71		B4		493.88	
72	73	C5		523.25	554.37
74	75	D5		587.33	622.25
76		E5		659.26	
77	78	F5		698.46	739.99
79	80	G5		783.99	830.61
81	82	A5		880.00	932.33
83		B5		987.77	
84	85	C6		1046.5	1108.7
86	87	D6		1174.7	1244.5
88		E6		1318.5	
89	90	F6		1396.9	1480.0
91	92	G6		1568.0	1661.2
93	94	A6		1760.0	1864.7
95		B6		1975.5	
96	97	C7		2093.0	2217.5
98	99	D7		2349.3	2489.0
100		E7		2637.0	
101	102	F7		2793.0	2960.0
103	104	G7		3136.0	3322.4
105	106	A7		3520.0	3729.3
107		B7		3951.1	
108		C8		4186.0	

As you can see from the funky chart that *for every octave you go up you double the frequency* and it is the same in reverse, for every octave that you go down, you halve the frequency.

Example: C4 is 261.63 Hz. To get to C5 we double the frequency so it is now 523.25 Hz. And if we wanted to go from C4 to C3, it would be 130.81 Hz. There, a few secrets to throw about.

Now let us create the tag for this whole sound thing.

I always imagine a wave as a 3 dimensional entity and with that I attach colours and size. So, for a low frequency wave I will think of it as a large and flowing wave with nice warm colours

like orange or deep red and the whole image is nice and slow. For higher frequencies I use smaller and faster waves and in harder colours like bright yellow or striking blue. This image is then enhanced further by having a person standing in front of the waves, usually me, but my name is Hertz and I am listening to these waves in a rent a car. Although this may now confirm the urgency for me to seek therapeutic help, it is the best way for me to remember things.

You can create whatever images or story lines to the definitions in this tutorial. They are your images and must work for you.

Next on the sound menu is **Amplitude**:
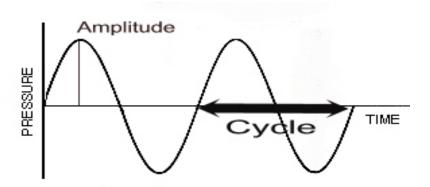
ii. AMPLITUDE

Generally speaking *this means the loudness or level of a sound or* **waveform**.

I prefer the word **waveform** for sound as it is the form or shape that the waves take and the further we go into this tutorial the more that term will make sense as waveforms vary in shape and character so, from now on, I want you to use the word **waveform** for sound.

 It is better defined with a simple graph. In fact, now is as good a time as any to introduce you to graphs. Enter **fig 2**.

Fig 2

As you can see, the waveform, it's actually a sine wave but don't worry about that for now, is 2 cycles and I have arrowed in the second cycle, no difference which cycle I arrow as they are both repeats. Anyway I had to arrow the second cycle so as not to intrude on the amplitude line in the first cycle.

The height or peak of the waveform is the amplitude and the length is measured as 2 cycles and this is done very simply.

Imagine a sound and how it starts. It starts from 0 then goes up, hangs about and then drops off.

In the diagram you can see the waveform starts at zero, goes up, drops to zero then goes to the negative area and then climbs to zero again. This is using the wave theory we defined earlier and all waveforms are represented like this, as a graph, and how each cycle behaves or how a number of cycles behave in relation to each other.

For now you do not need to worry about complex waveforms and any other factors regarding waveforms as we will deal with them as we go along, at your pace, that way you do not feel as if there is too much information to learn.

This is meant to be fun so let's keep it that way

The final most important component of sound is **TIMBRE.**

iii. TIMBRE

This is what defines the tonal quality of a sound.

A C4 note played on a piano and at the same level as a C4 note played on a saxophone does not produce the same sound or timbre. They are both the same level and both played at C4 but both have distinctly different sounds or timbres.

Timbres are made up of waveforms and it is these waveforms that go to make up the tonal quality of a sound. This is called timbre.

This is the main reason why one sound at a certain frequency will sound completely different to another sound at the same frequency.

When you come to use EQ you will understand why we do not have one overall frequency chart for all sounds.

A female voice at C3 will sound completely different to a male voice at C3 so the EQ properties will have to be different as the timbres of the two sounds are distinctly different.

The first challenge when EQing musical sounds is in deciding which area of the frequency spectrum corresponds to which element of a sound's timbre. If you want to emphasize the attack of a bass drum, where should you boost? Alternatively, if your vocals sound boxy, where can you cut or boost most effectively?

*This area is called the **bandwidth**, the area at which you want to boost or cut.*

It is also important to note that we *humans have a hearing range of between 20 Hz to 20 kHz*.

To be honest, anything at the 20 Hz level is felt more than heard. This is why producers cleverly add lower frequencies to a track for that big club feel.

5. Fundamentals and Harmonics

One thing that can help a little in deciding this is to know what frequencies correspond to the **fundamentals** of each musical pitch. For a start, this allows you to define the lower limit of the range of frequencies generated by pitched sounds.

Let's talk a little about **Harmonics**.

First, the emotional definition or description:

All musical tones have a complex waveform, made up from loads of different frequencies.

All sounds are formed using a combination of sine waves at varying frequencies and amplitudes.

Now it gets a touch more involved. If we look at the frequencies of a complex waveform, then the lowest frequency is called the **fundamental frequency**.

The fundamental frequency determines the pitch of the sound. The higher frequencies are called **overtones**. If the overtones are multiples of (x1, x2, x3 etc) the fundamental frequency then they are called **harmonics**. The overtones or **upper partials** as some people like to refer to them as, must be multiples of the fundamental to be known as harmonics. These frequencies and their amplitudes determine the timbre of a sound.

Now, the simpler explanation:

If you have a waveform that has a fundamental frequency of 100 kHz, then the second harmonic will be 200 kHz and the third harmonic will be 300 kHz and so on……
If you think about the irregular waveform of noise then you will understand that it has no harmonics. Noise contains a wide band of frequencies and it is generally accepted that, at waveform level, there are no harmonics as the waveform is non-repeating.

I have found that by boosting sounds below their fundamental frequency, noise of some sort is always introduced into the mix. In fact, it is always a good policy to use a high pass filter along with the use of EQ on a sound. I will come to this later when dealing with methods and techniques of using EQ.

The concept of harmonics, fundamentals, overtones etc seems daunting, but it's not. The above is simply to give you a better and more thorough insight into what sound and all of it's components are.

I do not expect you to know all of the above but if you want to be more proficient at your vocation, then it helps to try to understand what we have covered here.

However, as we have covered corrective EQ, albeit briefly, it is also vital to understand that EQ is not only about boosting and cutting frequencies, but also about **perception**.

Below is a list with approximate figures for instrument frequency ranges and their fundamentals. As I said earlier, I do not like lists that purport to be an accurate frequency range table for varying instruments, due to their inaccuracy. However, this list is simply to help you understand the fundamental and harmonic frequencies of a selection of instruments.

I am only listing the types of sounds that I feel will be used in most modern recordings for popular music, so have omitted the orchestral and acoustic instruments.

Instrument	Fundamentals	Harmonic
Kick Drum	30-145 Hz	1-6 kHz
Snare Drum	100-200 Hz	1-20 kHz
Cymbal	300-580 Hz	1-15 kHz
Acoustic Bass	40-295 Hz	1-5 kHz
Electric Bass	40-300 Hz	1-7 kHz
Acoustic Guitar	82-988 Hz	1-15 kHz
Electric Guitar	82-1319 Hz	1-15 kHz
Piano	28-4196 Hz	5-8 kHz
Bass Voice	87-392 Hz	1-12 kHz
Alto Voice	175-698 Hz	2-12 kHz
Soprano Voice	247-1175 Hz	2-12 kHz

The above are only guides, and to be used purely as a reference.

6. To Cut or to Boost

Another common problem that beginners make is to boost frequencies in a mix to try to make it stand out more. This is true particularly for drum sounds. The problem here is that noise is also boosted so what you end up with is a noisy sound in the mix, which stands out even more than intended. So the beginner tries to compensate by boosting other sounds to attain an even balance. Yep, you guessed it, more noise.

The true art of EQ is to cut, not boost.

However, boosting certain frequencies by small amounts can have a distinct impact on a mix.

The trick is to know when to cut and when to boost.

To be able to instinctively know when to cut (roll-off) or boost, you need to understand frequencies. Basses and pads sound great and full on their own, but combine them in a mix and they start to sound muddy. This is because they share so many of the lower frequencies. By cutting a certain range of frequencies from either or both, you will, in effect, give the perception of boosting the lower end, or upper end if you are cutting the lower frequencies. It always helps to cut certain frequencies with the aim of accentuating other frequencies. This is the *Art*.

Why?

Because the individual sounds sound clear and balanced, instead of muddy and biased. By taking away, we give. By boosting too many frequencies and channels, we compromise the headroom of the dynamic range of the audio. Whether it is in a mixer or DAW system, boost too much and you enter clipping and distortion territory.

The one area that is always the hardest to learn is the mid-end. So many sounds share this frequency spectrum that it can lead to confusion when it comes to finding a good balance of frequencies in a mix.

This is why cutting or boosting small amounts around distinct frequency spectrums of individual sounds can be so much more effective than choosing an overall frequency range for a number of sounds and altering that.

The same is true of the whole mix itself. EQ can be used to alter the apparent loudness of a mix, a technique commonly used by mastering houses.

Our hearing system's frequency response changes with loudness. It is important to monitor at a realistic and sensible volume when applying EQ, because the perceived effect may be quite different at higher or lower listening levels.

However, the fact that the ear gets more sensitive to high and low frequencies the higher the intensity of the sound, means that the brain tends to interpret any sound which is comparatively rich in these frequencies as loud.

Therefore, if you want music to sound louder at low listening levels, then it makes sense to boost at the extremes of the frequency range. This is what some hi-fi systems do when the Boost or Loudness button is used.

In most playback systems, you will see what is referred to as the 'smile' curve on graphic EQs.

The smile shape denotes that there is a dip in the mid frequencies and small boosts in the extreme frequencies. Always be aware as to the difference between actual and perceived.

It is also true to state that *at low listening levels*, the human hearing system encounters difficulties hearing very low and very high frequencies.
This is called the *Fletcher Munson Effect.*
In this instance, EQ is used to cut and boost selected frequencies, so that a more balanced gain structure is kept right throughout the hearing spectrum at low listening levels.

Masking is another problem that can be treated with sensible use of EQ.

How many times have you used a sound, that on it's own sounds excellent, but placed alongside another sound, gets swallowed up by the second sound?

This happens because the two sounds have very similar frequencies, so one 'masks', or hides, the other sound. This results in the masked sound sounding dull, or just simply unheard. EQ is a very good tool to use in these instances. By cutting away certain frequencies from one of the sounds, you will invariably expose and boost the frequencies of the other sound, thus accomplishing separation and distinction between the two sounds.

Another pitfall that most beginners and some pros fall into is what I call the EQ Syndrome.

This happens when a mix is poorly recorded with little separation in the sounds and EQ is used to try to 'separate' and 'cleanse' the sounds. This always results in a brittle mix with individual sound components sounding as if they do not belong together. Another example of this EQ Syndrome is when an engineer or producer feels that they have to EQ every channel to gain a stamp on their 'feel'. This complaint is quite common in certain Hip Hop and Rap songs whereby the drive (drum beat and bass line) of the song sounds separate from the vocals. This can actually be a good thing, if the effect is intended, but the ear begins to attune itself to the separate frequency bands instead of a rich tapestry of frequencies, and the song then starts to 'tire' the listener.

In these instances it is crucial to have a clean and balanced recording, so that the EQ process can be creative as opposed to corrective. You should strive to record the source sound elements at near enough the frequencies that you will eventually release. That way you will only need to make minor adjustments instead of sweeping corrections.

The added advantage of a clean and balanced recording, with emphasis on the correct source frequencies, is that you can always come back and remix the entire recording off a blank palette. The recording will never tire itself and never be constricted in terms of frequencies. This will always allow for refreshing remixes.

EQ has many guises.

We use corrective EQ in editing suites and production houses to isolate and diminish a frequency or sound, to accentuate or elevate a particular sound, frequency or recording.

Classic cases are that of broadcast engineers having to isolate the narrative or spoken part amidst a plethora of other background noises, or to simply remove a click or unwanted sound, and, even more commonly, to alter the spoken part to sound more pleasing in the event that it is harsh sounding. The latter is more in the domain of creative or musical EQ.

Using EQ as a tool to separate tracks is another favourite of producers.

This is actually quite an important procedure, but one that does need a careful approach.
It is imperative that your recordings are as clean as possible and have a perfect S/N ratio. This will ensure that small amounts of EQ boost, on selective frequencies, will attain the best results.

A lot of beginners make the mistake of poorly recording the source material and then using EQ to try to separate and boost the gains of the recorded tracks.
Separation can only be truthfully affected if there is a clear distinction between frequencies, so that you only need to isolate small frequency ranges and apply nominal boosts. Having a muddy mix of low end instruments makes for having to perform some extreme cuts/boosts, and this will be more destructive than creative.

Distance has a dramatic effect on sound.

High frequency sounds are **dampened** and absorbed by the friction of air, so they sound quieter or further away.

The further the sound travels, the more it is dampened.

We can mimic this in a mix by cutting backing vocals at, for example, the 10 kHz range, thus making it sit back in the mix and in turn bringing out the lead vocals to the front of the mix. Perception gained by clever use of EQ.

It then follows that if you dampen a sound, you will invariable give it the perception of being further away or quieter.

Use this piece of information.

If you need to bring a sound down in a mix, sometimes all it needs is to have the top frequencies rolled off.
The same thinking can be applied when you want to bring a sound out in a mix. Instead of boosting the gain of the whole sound, it can be very effective to boost some of the higher frequencies. But, remember that we are talking about very small changes here, not huge knob turns.

Another area of separation that is very important is that of *redundant frequencies*. This is my very flash way of saying 'frequencies that are not needed'.
You will find that there are a lot of instruments that share low frequencies, not just basses and kick drums. So, removing predefined low frequency ranges from some of the low end instruments in your mix can actually separate and define the low frequencies even more.

Recording your tracks rich in frequencies allows you the scope to cut or boost any frequency range, as it already exists in the recorded audio. I cannot this enough.

Separating the frequencies of instruments by the use of EQ is a traditional, yet subtle, method of creative and corrective EQ.

Creative EQ is an art form in itself.

Examples of this would entail, bringing out the best in a lead vocal line and yet keeping it balanced with the backing vocals, or to mix the drive element of a track to it's optimum club feel, or simply to use coloured EQs to add to or alter an existing sound. The list is endless. You are only limited by your knowledge on the subject, and, of course, having an ear helps, but this is not a pre requisite.

Finally, we use EQ at the mastering stage to best represent the final stereo mix for it's genre and medium. Once all the elements are in place and a mix of the session is handed to the mastering house, the real treatment takes place to bring out the best in the final mix and to make sure that there is a good dynamic range and all the elements are in place for whatever the market the mix is aimed at.

We have also arrived into the preset based EQ world for most hi-fi owners.

In fact, this has got to the point whereby hi-fi manufacturers put preset EQ settings on their systems for the listener to choose from. Ghastly presets called Pop, Ambient, Disco etc. are predefined EQ templates that you can tweak to your heart's content.

A well mastered mix will not need any additional EQ manipulating at the listening stage, as good mastering houses will treat the signal for optimum use on all listening mediums.

A good mastering house/engineer can make or break your track in the commercial vein. Their most valued weapon: EQ.

So, we now understand how important EQ is and the fundamental uses it might have, but we have not delved into the different type of EQ available.

7. Terminology and definitions-Filters

To further understand the terminology used in this E-book, I feel it is essential that you understand the following:

Cut-off frequency
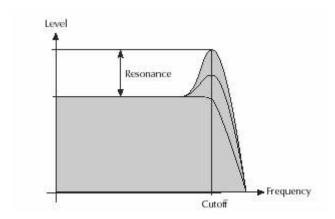
This is the point (frequency) at which the filter begins to filter (block or cut out). The filter will lower the volume of the frequencies above or below the cut-off frequency depending on the type of filter used.

Attenuation

This '*lowering of the volume of the frequencies*,' is called **Attenuation**. In the case of a low pass filter, the frequencies above the cut off are attenuated. In the case of a high pass filter, the frequencies below the cut off are attenuated.

Resonance

Boosting the narrow band of frequencies at the cut-off point is called resonance. Also know as Q and bandwidth, in effect, he higher the resonance, the narrower the bandwidth.

A very cool way of understanding what resonance sounds like is to perform what we call a '**sweep**'. Yes, another flash and funky term we programmers use to explain something really simple.

Sweeping the filter means manually turning the resonance knob, clockwise and anti-clockwise. Select a waveform, set the cut-off point and turn the resonance knob and listen to the results. As you are sweeping, the resonance goes through all the different frequency harmonics, of the waveform, and boosts/cuts them, at the cut-off point.

Q

Also known as '**width of the filter response**', this is the '**centre frequency**' of the bandwidth and is measured in Hz. Also know as bandwidth and resonance.

A high Q value denotes a narrow filter width (bandwidth). A low Q value denotes a wide filter width (bandwidth).

This is actually a very important piece of information because with the Q control alone, you can make your audio sound high and brittle or warm and musical. This does not mean that

you must use low Q values all the time, in the hope of attaining warmth, but you must understand what frequencies need filtering. If your intent is to use EQ as a musical tool, then be aware of what the Q value can do to audio. For creative EQ, this is a weapon often ignored.

8. Types of EQ

*To begin to understand the EQ, we need to first define the two categories it falls in, **Passive and Active***.

Passive EQs

These types of EQs have the distinction of being extremely simple in design and, more importantly, *they cannot boost frequencies*, only cut. The way they work is actually very much to do with perception.

By cutting, for example, low frequencies (bass), they make the mid and high frequencies sound 'louder'.

Passive EQs do have their uses. Although they are inflexible, they can perform reduction tasks reasonably well. By cutting high frequencies, they are able to cut or lower hiss (high frequency noise). However, by their very nature, passive EQs, or filters, have to then have the signal boosted to compensate for the cut. This, in itself, introduces noise into the signal path. The noise coming from the amp used to boost the signal.

Active EQs

Because of the limitations of passive EQs, most EQs are built around active filter circuits which use frequency selective components, together with a low noise amplifier.
And it is this type of EQ that we are now going to concentrate on.

Fixed Frequency EQ

Pretty self explanatory, *this EQ allows cut/boost of one or more frequencies*. There are no additional controls over the usual components, like bandwidth, Q, etc.

Peaking EQ

A peaking EQ is an EQ which boosts a specific band of frequencies.

Whereas a shelving filter has a shelf like curve, this filter has a bell shaped curve. The Q setting determines the width of the bell, while boost or cut determines the height or depth of the bell.

Two Band or Three Band

These types of EQ simply have two or three separate frequency ranges. Usually denoted as low, mid and high, these bands can only be cut or boosted.

Shelving Filter/EQ

We have touched on the use of tone controls that are forms of EQ. These controls control a type of filter that is called a **shelving** filter. In the case of the bass and treble knobs, low pass and high pass shelving filters are used respectively.
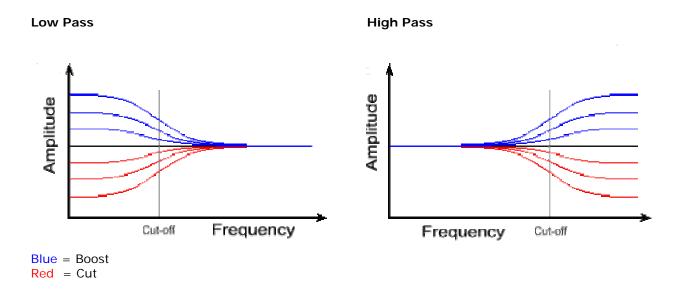
*A low-pass shelving filter passes all frequencies below its cut-off frequency, but **attenuates** all frequencies above its cut-off frequency. Similarly, a high-pass filter passes all frequencies above its cut-off frequency, but affects all frequencies below its cut-off frequency.*

This is the simplest type of active EQ. This EQ can shape response in a number of ways: boost/cut low frequencies, boost/cut high frequencies. This is why I have included the graph to demonstrate what happens with the filters, low and high pass, in this type of EQ.

Most mixers will allow for low and high frequency EQ, and in the case of shelving filters, their mid frequencies are usually fixed.

It is also common for the filter slope to be 6 dB per octave. This allows for a gentler effect. The shape is shelf like, so the boost or cut is progressive over a range. Filters do not have a no-effect at a frequency and then instantly jump and suddenly reappear at the next frequency. They have to get there somehow. The way, and by how much, they get there is called the **gradient** or **slope**. In the case of the shelving filter, the most common slope is 6 dB gain change per octave (doubling of the frequency). It takes time for the filter to attenuate frequencies, in proportion to the distance from the cut-off point. This is the **slope**.

The diagram below illustrates what happens if you cut or boost frequencies in a low-pass and a hi-pass filter.

Low Pass   **High Pass**

Blue = Boost
Red = Cut

Graphic EQ

A graphic equalizer is simply a set of filters, each with a fixed centre frequency that cannot be changed.
The only control you have is the amount of boost cut or in each frequency band. This boost or cut is most often controlled with sliders. The sliders are a graphic representation of the frequency response, hence the name 'graphic' equalizer.
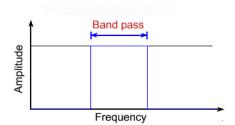
The more frequency bands you have, the more control and accuracy you have over the frequency response.

Mixing consoles rarely have graphic EQs, but PA mixers often have a stereo graphic EQ for EQing the final stereo output.

A graphic equalizer uses a set of **band-pass** filters that are designed to completely isolate certain frequency bands.

The diagram below shows the frequency response of a band-pass filter.

Band Pass Filter

*A filter that passes frequencies between two limits is known as a **band-pass filter**.*

This is a great filter. It attenuates frequencies below and above the cut-off and leaves the frequencies at the cut-off. It is, in effect, a low pass and a high pass together. The cool thing about this filter is that you can eliminate the lower and higher frequencies and be left with a band of frequencies that you can then use as either an effect, as in having that real mid range type of old radio sound, or use it for isolating a narrow band of frequencies in recordings that have too much low and high end.

Try this filter on synthesizer sounds and you will come up with some wacky sounds. It really is a useful filter and if you can run more than one at a time, and select different cut-offs for each one, then you will get even more interesting results. Interestingly enough, band pass filtering is used on formant filters that you find on so many softsynths, plugins, synthesizers and samplers. Emu are known for some of their format filters and the technology is based around band pass filters.

It is also good for thinning out sounds and can be used on percussive sounds as well as creating effects type of sounds.

I often get emails from programmers wanting to know how they can get that old radio effect or telephone line chat effect or even NASA space dialogue from space to Houston. Well, this is one of the tools. Use it and experiment.

You will enjoy this one.

Notch Filter – also know as Band Reject Filter

The inverse of a band pass is the notch filter.

This is a very potent EQ/filter. It can home in on a single frequency band, and cut/boost it.

Used specifically for 'problem' frequencies, the notch can be one of the most useful filters. This is the exact opposite of the band pass filter. It allows frequencies below and above the cut-off and attenuates the frequencies around the cut-off point.

Why is this good?

Well, it eliminates a narrow band of frequencies, the frequencies around the cut-off, so, that in itself is a great tool. You can use this on all sounds and can have a distinct effect on a sound, not only in terms of eliminating the frequencies that you want eliminated, but also in terms of creating a new flavour to a sound.
But its real potency is in eliminating frequencies you don't want. Because you select the cut-off point, in essence, you are selecting the frequencies around that cut-off point and eliminating them.

An invaluable tool when you want to hone in on a band of frequencies located, for example, right in the middle of a sound or recording. I sometimes use a notch filter on drum sounds

that have a muddy or heavy mid section, or on sounds that have a little noise or frequency clash in the mid section of a sound.

Parametric

This filter controls three parameters, frequency, bandwidth and gain. You select the range of frequencies you want to boost or cut, you select the width of that range and use the gain to boost or cut the frequencies, within the selected bandwidth, by a selected amount.

The frequencies not in the bandwidth are not altered. If you widen the bandwidth to the limit of the upper and lower frequencies ranges then this is called shelving. Most parametric filters have shelving parameters.

Parametric filters are great for more complex filtering jobs and can be used to create real dynamic effects because they can attenuate or boost any range of frequencies.

Basically, the parametric EQ places several active filters across the frequency spectrum. Each filter is designated to a frequency range, low, mid, high etc. You have the usual cut/boost, resonant frequency and bandwidth. It is these qualities and the control over them that places this particular EQ in the producer's arsenal of dynamic tools, and makes it detailed and versatile.

However, you need to understand what you are doing when using a parametric EQ, otherwise things can go very wrong.

Understand frequencies and sound, and you will be in total control.

Quasi-parametric EQ

This is just another form of parametric EQ but without the bandwidth control.

Sweep EQ

This is very similar to a band pass filter, but with variable centre frequency, and no control over the width of the filter response (Q).

You will find that most mixers will have band pass EQ, and some will have sweep EQ (where the centre frequency can be varied, also known as '**tuneable**'), but very few, mainly digital, will have parametric EQ.

Paragraphic EQ

Another variation on the graphic EQ. This EQ provides control over the centre frequency of each band.

9. Phase the bi-product

A very important aspect of EQ, in relation to affected and non-affected frequencies, is that of phase.

We know that affecting the frequencies that we have chosen for equalisation, also affects the phase of those selected frequencies, in relation to the unaffected frequencies. The process itself also affects the frequency response of the signal being treated. We are talking about tiny offsets here. Every time a frequency range is selected and treated, the affected frequencies will exhibit displacement, in relation to the unaffected frequencies. This offset is phase. Whereas we are not talking about big swirling phase effects, as in guitar phasing, we are, however, talking about the pure definition of phase. This is probably not something that you will hear as phase, but it is something that affects our perception of the treated frequencies.

Depending on the nature of the displacement, we perceive the treated frequencies as distance.

Why is this important?

This is what differentiates the tonal characteristics of analogue hardware and digital software EQs.
The analogue EQ unit will exhibit far more musical phase changes than it's digital counterpart, and at very low gains, whereas the digital EQ unit will have the advantage of leaving the phase relationships hardly affected, thus allowing for more robust gain changes.

They both have their uses.
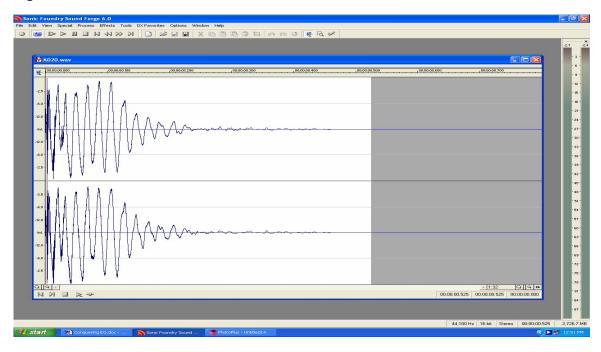
10. Let It Begin

We have covered EQ as a subject. It is now time to begin with walk through examples.

The following examples will deal with instruments and vocals. This way, we have a good variety of scenarios to deal with, and a good range of frequencies to cover and manipulate.

Drums – Kick

Fig 1
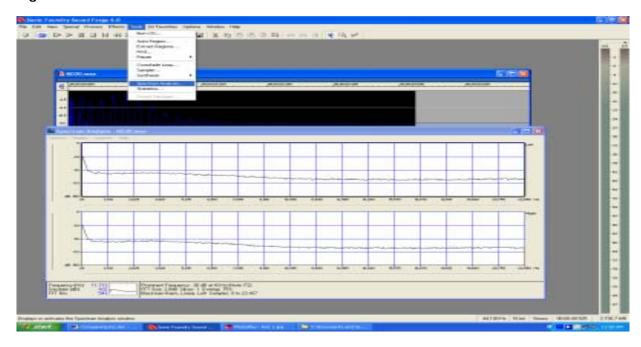
Listen to this file (**K020**)

As you can see from the above image (**Fig 1**), I have opened up a kick file in Sound Forge. We are now going to manipulate this file and create new files, by the simple use of EQ.

However, before we perform any type of manipulation, I want to show you how to evaluate and see the frequency spectrum (range) of any selected audio file.
For this, we use a piece of software, or hardware, called a spectrum analyser.
Sound Forge has this tool under the menu option, *Tools - Spectrum Analyser.*

The image below (**Fig 2**), clearly displays the frequency spectrum of the kick when played.

Fig 2

By using the analyser we are able to fully see the frequency start, the frequency body and the frequency tail-off.
This type of tool is invaluable in helping us to apply EQ.
You will appreciate how useful this tool is when we come to removing unwanted frequencies,
or when we come to using EQ to either correct flawed vocals, or thin out or thicken vocal lines.

The Y axis (left hand side figures) denotes amplitude/level in dB.
The X axis (bottom figures) denotes the frequency in Hz.
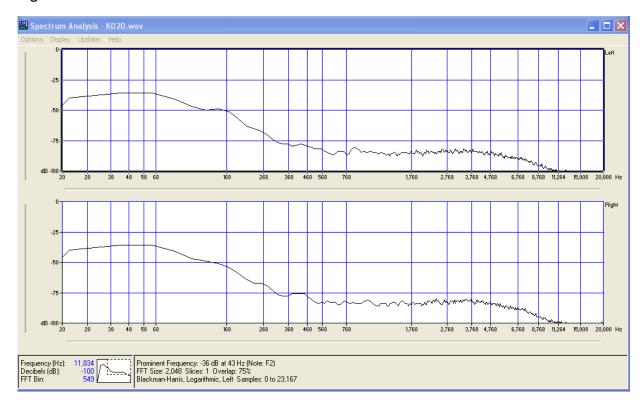
The analyser tells us that this kick starts on a frequency of 20 Hz at – 45 dB (attack), rises to
65 Hz at -36 dB (height of attack's decay), drops to and levels off at around 88 dB.

This helps us a great deal in understanding the characteristic and shape of the kick file and
this in turn gives us a detailed numerical and graphical reference for setting up our EQ
parameters.

What did I just say, in English?
Simple: check the same image but zoomed in (**Fig 3**)

Fig 3

I highlighted the peak of the kick file's waveform with the mouse and it gave me the figure above.

I can move my mouse over any part of the waveform and it will highlight the data I need.

With this example, I can now decide what type of EQ I want to apply and by how much.
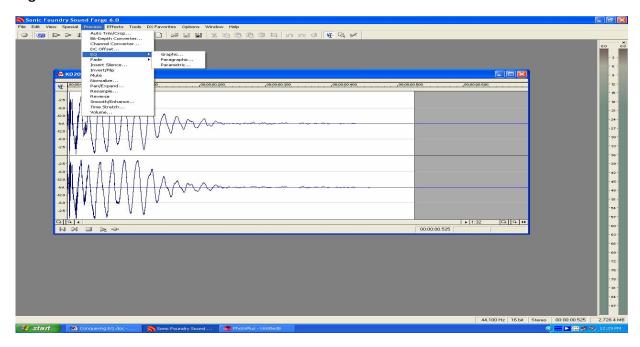
The *punch component*, or attack, of most bass drums lies between about 60 and 110Hz. This is where you find the low-end energy of most kicks. Below this range, you'll mostly feel, rather than hear, any boost, and it's easy to either neglect this and be left with all sorts of frequency imbalances in your mix, or to be confused by what is actually the 'bottom-end' of your mix..

With sounds that tend to be '*warmer*', as opposed to '*cutting*', you will invariably find that the frequency area concentrated on is between 180 – 220 Hz.

Working on a kick that might need to be more prominent in the mix, or cut through on mid range monitors, the 2 – 6 kHz range is where the manipulation takes place.
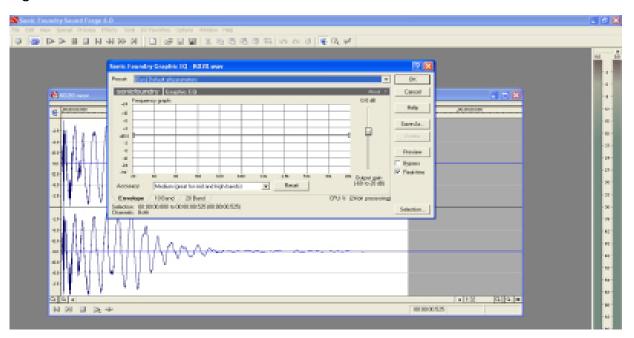
I am going to choose an EQ (**Fig 4**), graphic in this instance, from the menu option *Process – EQ – Graphic.*

Fig 4

And here is the Graphic EQ (**Fig 5**).

Fig 5

I have not input, or drawn, in the EQ curve. The Graphic EQ is currently at default with all the parameters at 0.
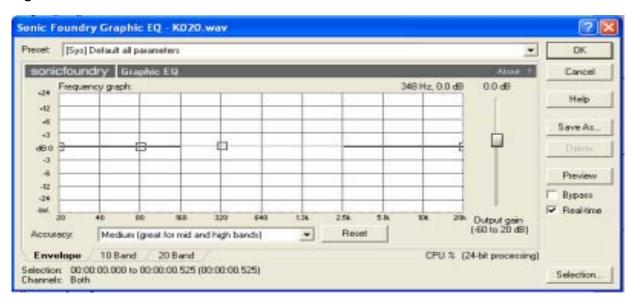
Now, let us shape the EQ curve by using the nodes in the Graphic EQ window.
The default line (middle of screen) is at 0dB, across the whole frequency spectrum.
By placing the mouse over any part of the line, I can create a '*node*' and then move this node.
I can keep on creating nodes, so that I have control over the default line. The more nodes, the more detailed your control over the shape.

I have always preferred visual interfaces for these types of dynamic manipulations, as opposed to inputting fields and numerical data into graphs. I can visually create the EQ curves here, instead of having to input numbers and hope for the best.

There is also something very satisfying in having such instant and visual control.

Fig 6 clearly shows what I mean by nodes.
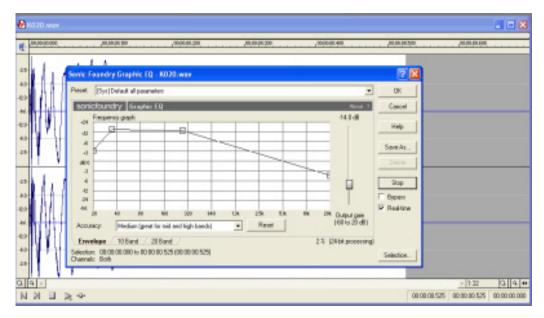
Fig 6

As you can see from the above image, I have a total of four nodes (*little boxes*) across the default line.

I am now going to move these nodes around and create an EQ shape for the kick drum file
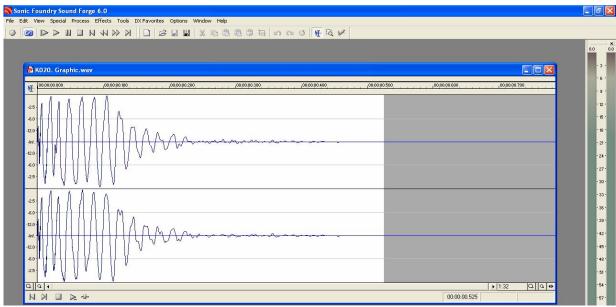
Fig 7

And once you press 'OK', that will render the new EQ shape over the audio file.
The result is below (**Fig 8**).

Listen to result (**K020 Graphic**)
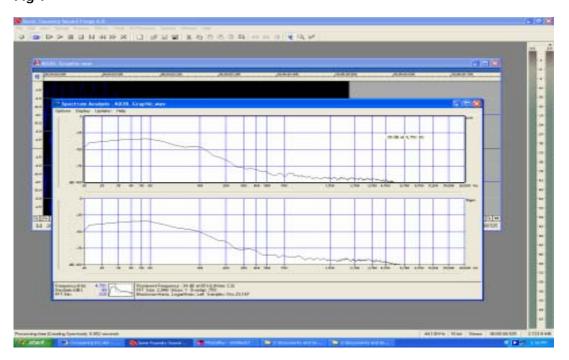
Fig 8

The kick file now sounds deeper, with less 'top end'. The EQ shape accents the attack and decay of the attack, and drops rapidly from there thus making the body and tail seem both quieter and as if the higher frequencies have been filtered, which they have.

Now let us put this file into the spectrum analyser. This will clearly show if I am right or wrong (**Fig 9**).

Fig 9

Yep, spot on.
You can see that the attack and attack decay are peaked and smoothed between 48 Hz – 66 Hz. After that, the amplitude drops dramatically over the frequency spectrum.

Whereas for now the spectrum analyser is useful, it is however not essential. Later, when we come to deal with isolating frequencies or trying to use EQ as a means for a better mix, it will become invaluable.
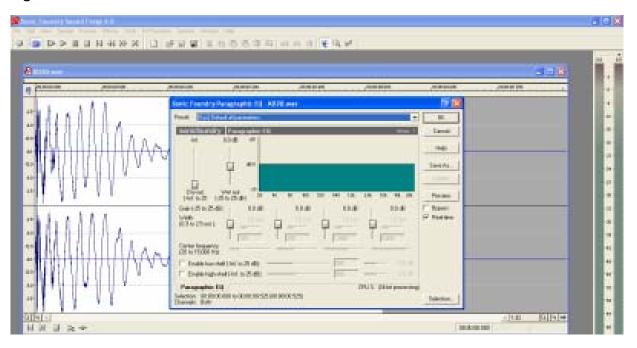
Have fun with this tool feature in your audio editor. You will be amazed at how much you will learn, just by viewing frequency spectrums of different audio files.

Now let us use the same kick file, but with a different type of EQ, an EQ we discussed earlier, the Paragraphic EQ.

So, to begin, we perform the exact same sequence as before. In fact, it is good practice to stick to a procedure and to keep practicing that. This helps you to perform tasks quicker and to fault find in the event that you have made a mistake in the process.

Have a look at **Fig 10**. It shows the parameters of the Paragraphic EQ.

Fig 10

There are 4 bands and each bandwidth can be adjusted. This Paragraphic EQ has a little more control than the usual Paragraphic EQs, in that you can not only *shape the bands by adjusting the centre frequencies and their amounts, but also vary the position of each band.*

In **Fig 11**, I have chosen to remove almost all of the low end and boosting the mid to top end.

This has given me a kick with very little low frequency content, hard and thin sounding.
This is a good way of explaining how to use this EQ and how it sounds.

Listen to the Pargraphic EQ (**K020 Paragrahic mid to top**)

Fig 11
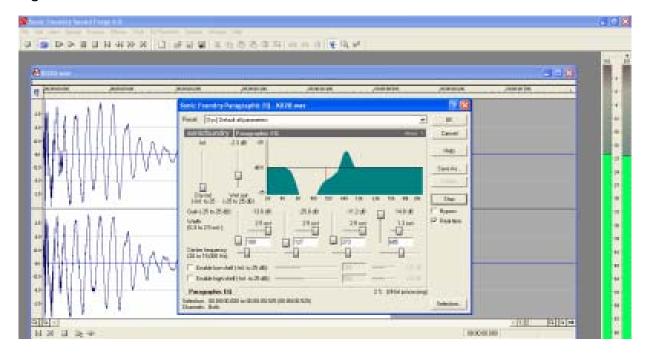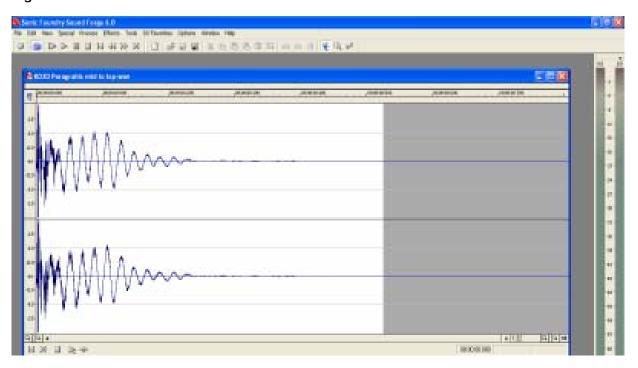
Fig 12 shows the new kick file after being rendered with the Paragraphic EQ.
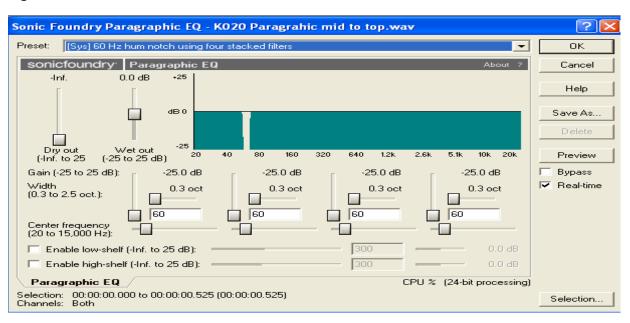
Fig 12

The new kick waveform (**Fig 12**) clearly shows what the Paragraphic EQ has done to the frequency and amplitude of the waveform.
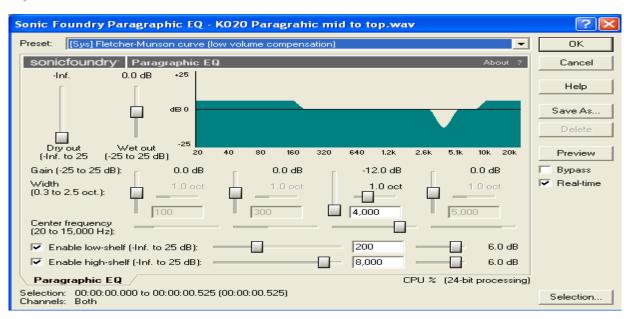
The Paragrahic can be used very effectively as a Notch Filter (**Fig 13**) and also to demonstrate the *Fletcher Munson Curve* (**Fig 14**).

Fig 13

The above shows how to use the Pargraphic EQ as a Notch Filter to isolate and remove 60 Hz cycle hum. This is a great tool to have as there are times when your audio file might exhibit mains hum at 60 Hz. This way you can isolate and remove the hum.
Cool huh?

Fig 14

Now let us use the same kick file, but this time we will use a *Parametric EQ.*

I like using this EQ, because not only is it simple to use, but it also has additional filter settings that make this EQ very versatile.
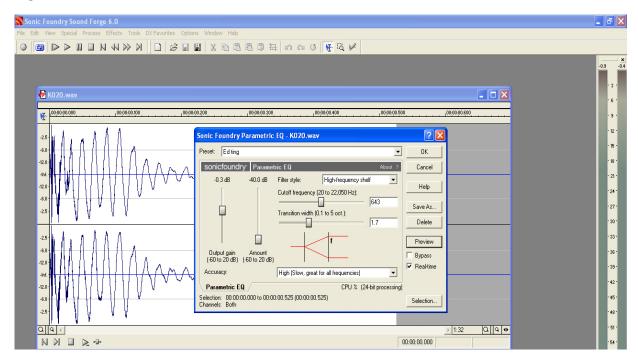
Look at the settings I am using in **Fig 15**.

I have adjusted the parameters so that I am left with a high frequency shelf. I am actually *cutting* frequencies and not *boosting.*

The kick file I am using for these examples is a good all round kick file, that covers most of the frequencies I would expect to see in a file of this nature, and is not limited to any particular frequency. That is why it is such a good file to use.

As we discussed earlier, cutting is always preferred to boosting, and this example reflects that thinking beautifully.

Fig 15

Now listen to the result (**K020 Parametric high shelf**).

And now I have used the *Band Pass Filter* on the Parametric, to give the kick sound a *boxy* feel (**Fig 16**).

Listen to 'boxy' kick (**K020 Parametric boxy**)

Fig 16
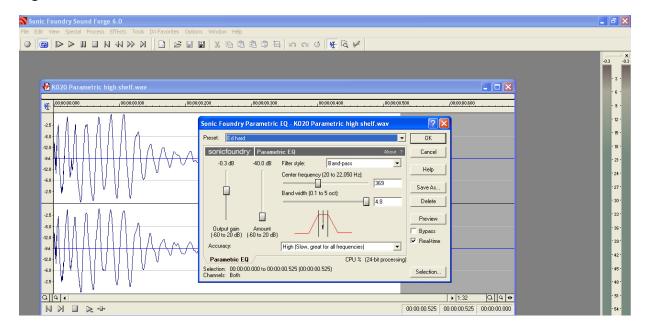
Fig 17 is an image of a 2 *band* EQ, as discussed earlier.

Fig 17

As you can see, you have only two nodes to play with. This EQ is really self explanatory.

For four bands of EQ, simply add another two bands to the above. The thinking is the same. Listen to the 2 band EQ kick (**K020 2 band EQ**)

Drums are probably the easiest instruments to EQ.

The real art of kick drum EQ is to consider the bass line sound in the song that the drums reside in. As they invariably share similar frequency ranges, they must be processed with care and attention.

Apply small amounts of EQ, as the process and result of applying any dynamics will result in some form of degradation. So, be sensible and wary of drastic changes. Keep things natural.

Separation also comes into the equation and the whole 'drive' of modern Hip Hop and Dance songs are centred round the marriage between the kick drum and the bass line.

11. Snare

The snare sound is equally easy to EQ. Snares tend to have a good frequency range as there are so many types of snares that to try to and classify them in one specific frequency band, would be naïve and uninventive.

However, the *low-end*, or fatness, of snares generally lies in the 100-440 Hz frequency range.

The *energy* of the snare's body will tend to be in the 750-1.4 kHz frequency range and the *crispness* of the attack will lie in the 4-8 kHz frequency range.
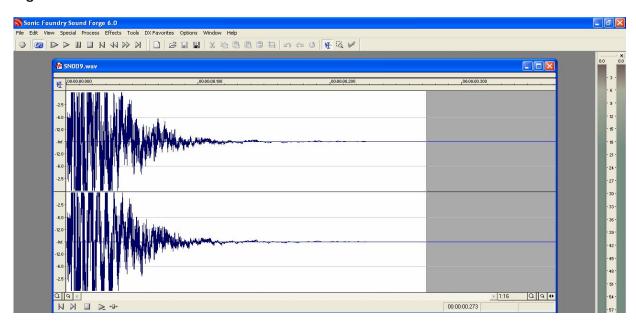
These are generalisations, but they will give you an idea as to where you might want to begin, when applying EQ.

But, as with all these examples, the Spectrum Analyser is your best friend.

With EQ it is often very easy to create a varied selection of snares from one *source* snare.

Let us begin with a simple snare sound (**Fig 18**).

Fig 18

Listen to this snare (**SN009**).

A standard mid range bucket snare, comprising a metallic base with some noise element. Generally, a good Hip Hop snare sound.
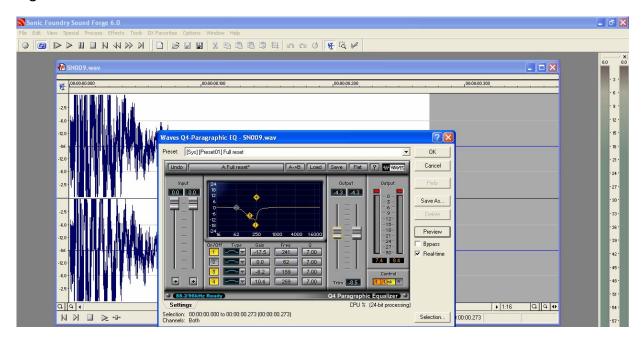
Now let us create another snare from this *source* snare.

I am going to use a 4 band Paragraphic EQ, but this time I am going to select the type of band I want to adopt.

With certain software/hardware dynamics, you can have the facility to select the filter shape, or EQ shape, of the band.

In this instance I am using the Bell shape. You can see from **Fig 19**, I have selected the Bell shape for three of the bands. The Bell simply denotes the shape of the frequency curve.

Fig 19

With the settings I have used above, I have concentrated more on the mid-range frequencies of the snare sound.

This is a good way to remove the low-end frequencies and keep the mid-range '*alive*' and the top-end as is.

You can clearly see what frequency bands I have used and by how much.

This EQ allows me to move the bands via the numbered nodes. I can also turn off any nodes that I do not want to use. I can also change the shape of the bands too (**Fig 20**).

This gives me more flexibility and choice, and editing this way is so much easier and instant.

Fig 20
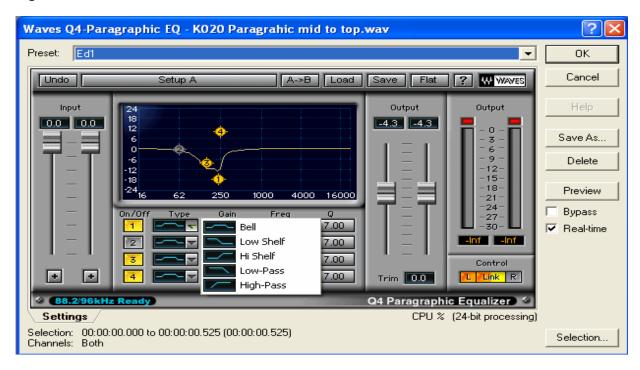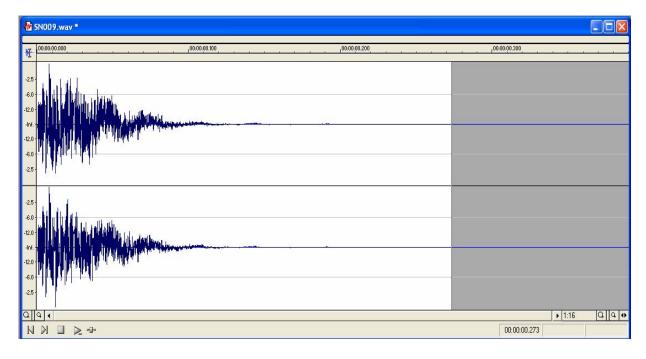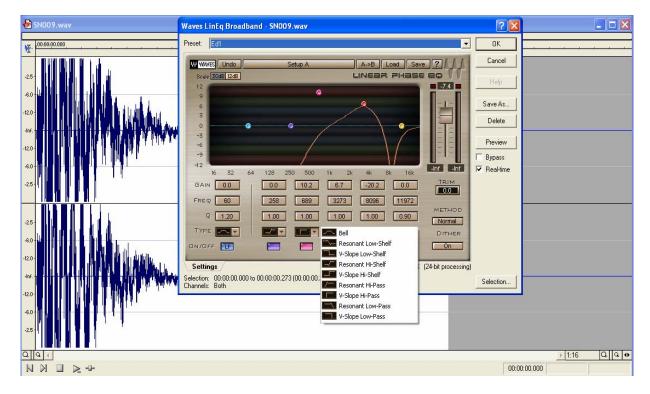
Fig 21 shows the result, after rendering the original snare with the EQ.

Listen to the new snare (**VSN009. 4 band EQ Waves**).

In **Fig 22**, I have selected a 5 band EQ.

As you can see, I can change the EQ shapes for each band, but I also have *Resonance* and *Q* for the filter types.
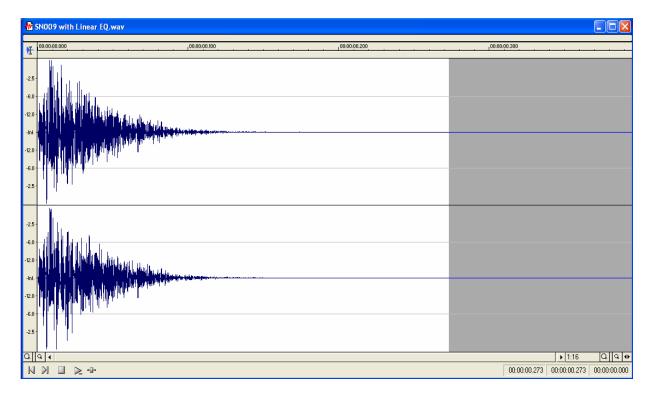
If you are having problems understanding the above, then go back to the earlier part of this e-book where *Resonance* and *Q* are explained.

I like this EQ a lot. It allows me far more detail and control over the frequency bands. I also have a main out for the bands, and with it's own parameters: frequency shape, resonance, Q, low-pass filter and overall gain.

Just for fun, I created a crazy shape for the bands, but it really sounds tight and painful.

This EQ will give you endless hours of fun and is one of the most detailed of the EQ plugins available.

This is the snare after processing (**Fig 23**)

Listen to the snare (**SN009 with Linear EQ**)

I could go on and on with examples of snare EQ, but I feel that you have enough information above to start to try some of your own EQ projects.

As mentioned earlier, and many times, keep it simple and do not apply too much EQ unless you are after a certain effect. Otherwise, less is always more, and in particular, when it comes to *corrective* EQ.

I picked kicks and snares as my subject matter here, simply because they are the most common percussive sounds that get treated with EQ.

It would be pointless to repeat the above for claps, hi hats, or any other percussive sound, as you have more than enough information to help you to apply these techniques to any percussive element.

If you do have any problems with frequency ranges, just use the Spectrum Analyser.

You know how to use it, so …….use it!

Let your ears be the most potent and trustworthy tools at your disposal.

12. Drum Loops

Drum loops are a little more complex as the loop itself might incorporate a number of percussive sounds that also need to be considered when EQ is applied. However, this is again not as complicated as it seems.

The trick here is to try to make sure that you treat each drum sound separately, and that the final stereo mix of the drum loop will incorporate varying EQ curves for the isolated drum sounds.

I find that using the same EQ software/hardware for all the individual drum sounds makes the final loop sound more natural. As mentioned earlier, different EQ units have different tonal qualities, so keeping to the same unit makes the process sound more natural.

There is nothing worse than hearing a loop that houses drum sounds that sound as if they are from different kits.

Keep it simple, keep it safe.

For this to happen, you will need all the drum components in separate tracks, and instances of the same EQ used on each track..

If, however, all you are presented with is a stereo file of the drum loop, with no individual tracks of the drum components, then you need to consider the whole frequency range of the entire drum loop when using EQ.

Always be aware of what it is that you are applying EQ to. You need to make sure that the treated sound has a natural finish to it.

With drum loops you need to think ahead, and even more importantly, to be aware of what other sounds will be used with the loop. If the drum loop sounds a bit thin, don't despair, the bass sound might compensate.

In other words, unless you are after a specific effect, *the drums must always have a good dynamic frequency range, with flowing frequencies*, as opposed to a compacted frequency range. This will ensure a lively drum loop with all the components nicely residing in their frequency ranges.

Used gently and cleverly, EQ can dynamically change a drum loop's sound but maintain the believable and natural dynamic frequency content inherent in the individual components of the drum loop.

Another area to be very wary of is that of *frequency crossovers* (overlaps). The high end of a snare sound might start to share the low end frequencies of a hi hat sound. The mid range of a kick might share frequencies with the low end of a tom drum, or vice versa.
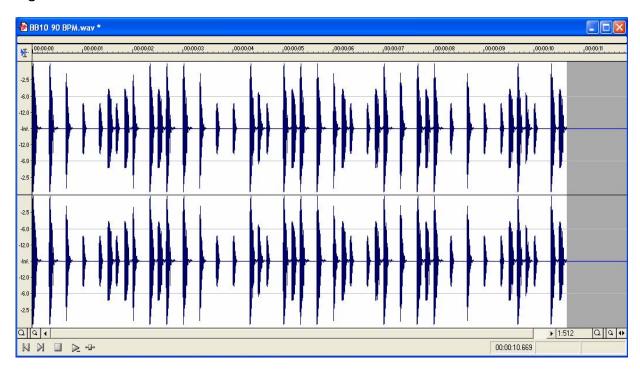
Always be cautious of these frequency overlaps, because if you are not, then there will be gain boosts where frequencies overlap, plus possible phasing issues and a host of other anomalies.

Let me explain the above with examples.

First off, we will use a si8mple drum loop and EQ that so as to give varying versions of the same loop.

Fig 24 is a straight forward drum loop incorporating a kick, snare and hi hats.

Fig 24



Listen to (**Beat 1 90 BPM)**

Now let us EQ it.
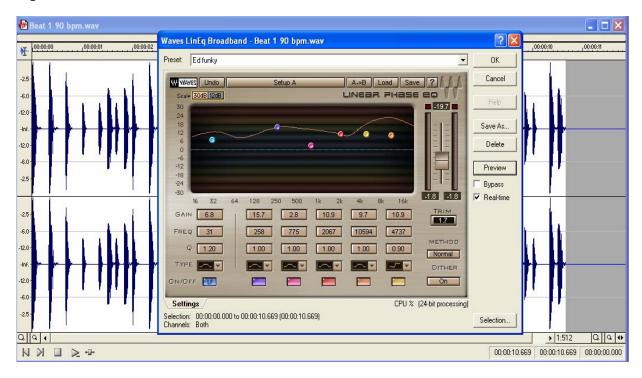
I am using the same EQ as I used earlier, the 5 band affair.

**Fig 25** shows what I have done with regards to shaping the EQ curves.
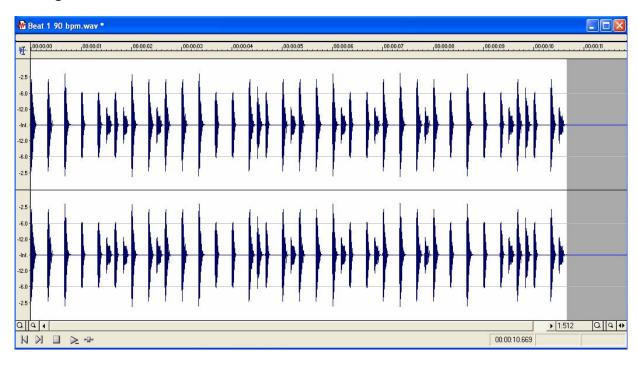
I have gone for a drastic separation effect. The drums now sound more separated and crispier, slightly dirtier, but louder.

This is not subtle, but I have deliberately done this so you can fully appreciate what we can do with EQ

Fig 25

And **Fig 26** shows the result.


Listen to this loop (**Beat 1 after Linear EQ**)
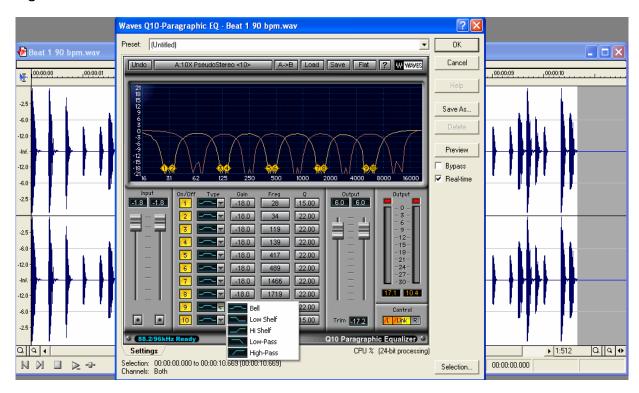
In this next example (**Fig 27**), I am going to show you how to fill up the frequency '*space*' (the area across the whole frequency spectrum) of the loop and mimic '*stereo spread*'.

I am using a 10-band Paragrahpic EQ. I need loads of bands so I can have greater control over all the individual frequency bands. As you can see from the image, the EQ also offers me all the usual band shape options.

What am I trying to achieve here? I am trying to create what we call '*Pseudo Stereo*'.

Pseudo Stereo usually means creating a stereo effect using a filter or dynamics. In this instance, we are using EQ to create a stereo effect.

Fig 27

Listen to the loop (**Beat 1 10 Band Para**)

This is quite an extreme example and, as you can hear, it is not perfectly balanced across the axis. This is more to do with the way the loop was constructed with the hi hats being panned just off centre.

However, I have deliberately made it extreme so you can see all the band points and how they affect the audio.
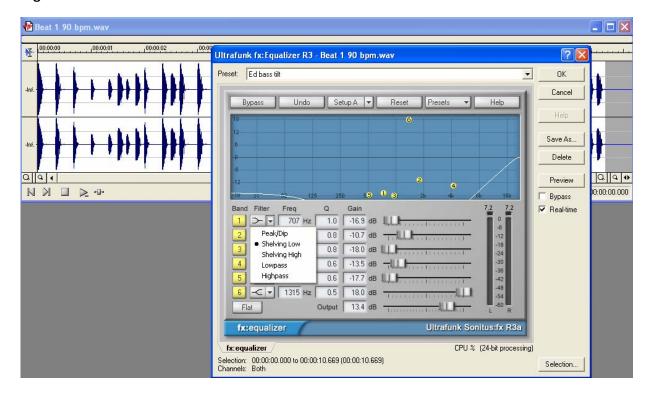
This next example (**Fig 28**) is a template I have created using a 6 band EQ with more parameters for '*Peak*' and '*Shelving*'.

The parameters are extreme and actually off the chart. I can't even see the nodes.

Why have I created this template? I have created this for drum layering purposes.

All these software based EQ units allow for saving your own EQ edits. It is a good policy to save every template you create. Calling up templates for any given project makes life so much easier and saves you loads of hard work trying to replicate a template that you created but have since lost. Imagine if you had created the greatest of all EQ templates, but didn't save it? It would make a grown man weep.

Fig 28

Now listen to this (**Beat 1 shelf bass tilt**), and tell me it is not fat as Hell.

This makes for a great loop that can be layered with another drum loop. The two together will make a really deep and crisp loop.
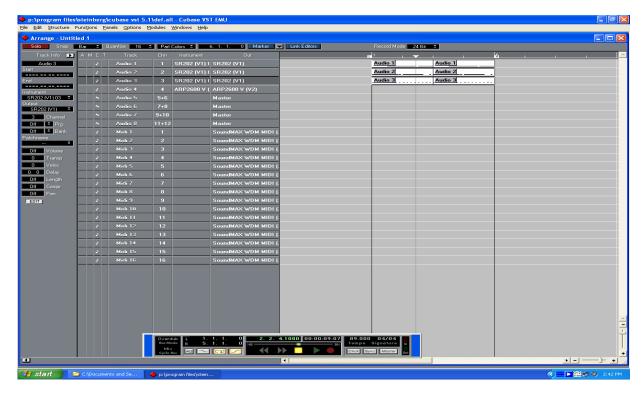
You can also do the exact opposite, creating a beat that is rich in mid to high frequencies, and then layer that with a deeper low-end beat. You can even play your own drum sounds over the beat and create a new beat. It is endless.

You are only limited by your imagination, and not by the tools.

We will now run through an example incorporating a drum beat with all the individual components on separate tracks in a software sequencer.

Fig 29 shows 3 track instances of audio, recorded in Cubase (sequencing software). The tracks are broken into the following drum sounds: snare, kick and hi hat.

Fig 29

Listen to the beat (**The Beat EQs**)

This is a very simple drum beat, and it will serve our purpose perfectly.

I have opened 3 instances of EQ, one on each channel.

The EQ units that I am using are the standard default 4 band EQ provided with the software.

I have treated the snare to be a little harsh and dirty, with a lot of mid to top-end.

The kick has been treated to make it thumping and pronounced in the low to mid-end.

The hi hats have been treated to be a little on the mid-end, so as to compliment the frequency gap between the kick and snare (**Fig 30**).

Fig 30

The 4 nodes, one for each band, are clearly evident in this image.

This is the EQ unit used for the hi hats. You can see that I have boosted the mid to high frequencies in a gentle manner, but with incrementing values.

You must always be very aware of the frequency ranges that you are boosting or cutting. Keep checking all the EQ settings and make sure there are very few overlaps, if any.

I have deliberately used this type of EQ curve for the hi hats, because I know that I will be treating the kick with more emphasis in the low to mid-end, and the snare with emphasis in the mid to top-end.

This method affords me a good frequency *spread* across the whole spectrum.
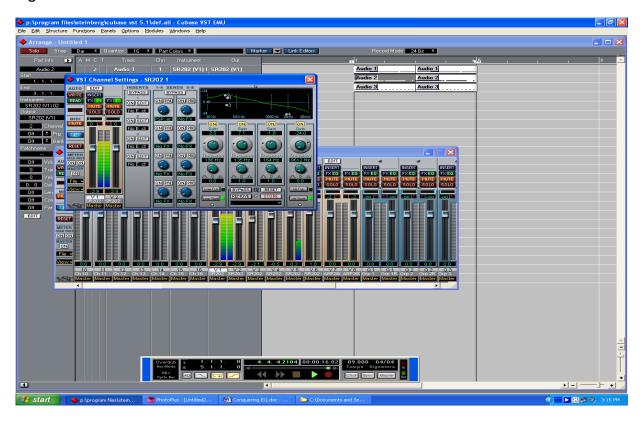
If there are serious overlaps in the EQ curves, then the file will suffer.

The only time I deliberately overlap frequencies is when I am after a very specific effect. An example would be boosting the mid range of a drum loop so as to fit the bass line in it's own space.

In this instance I would overlap the kick and snare frequencies and boost the shared frequency range, and then lower the gain so as not to cause any tonal anomalies like distortion, clipping etc.

Fig 31 shows the kick EQ treatment.

Fig 31

You can see the shape of the EQ is tailing off towards the mid-end.

I have done this so when the two EQ curves are put together, they form an even frequency boost across the whole frequency range of the audio file.

This has kept the current ranges open for the snare to explore the space I have reserved for it.
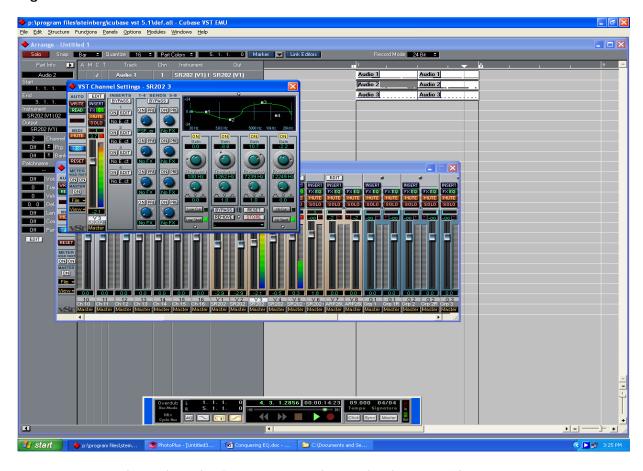
Again, you can always change the shapes of any of the EQ units.

The whole point of this exercise to show you how to sensibly select EQ parameters and make full use of the frequency bands available.

Bear in mind that we have the option to turn off any of the bands in this example. This allows us to use the EQ unit as a one band, all the way up to four bands.

Fig 32 shows the EQ curve for the snare sound.

Fig 32

As you can see, I have shaped quite a pronounced curve for the snare EQ.

I have removed almost all the low to mid-end frequencies, and boosted the mid to top-end frequencies, so as to accommodate the hi hat frequencies.

Now listen to how the finalised file sounds like here (**The Beats with sep EQs**).
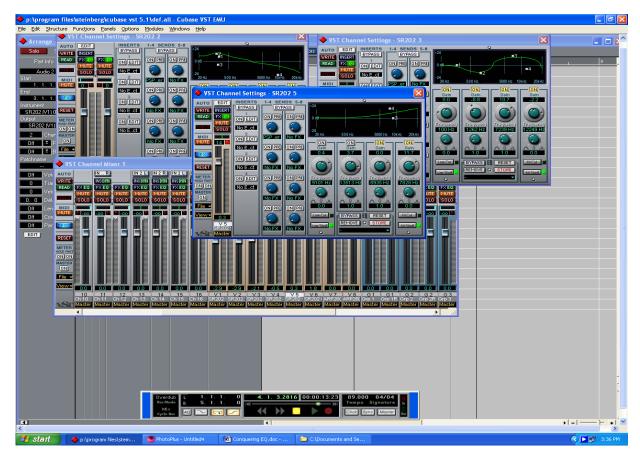
You can hear the differences from the '*dry*' (untreated) and the '*wet*' (treated) files.

It is clearly evident that the original beat has undergone treatment.

I have, of course, used extreme curves so as to show you how to get a good even level across the whole frequency range of a given file.

To conclude this section, take a look at the EQ curves of all three drum sounds in **Fig 33**, and you will notice that combined, they form a half decent even EQ curve.
Take note of where I have cut and where I have boosted.

Fig 33

The time has come for us to move onto the most confusing and difficult of all sounds to treat, **The Voice**.

13. Voice

This is THE area that most people have trouble with when it comes to using EQ.

For some strange reason, even engineers have to ponder a solution to an EQ problem where the voice is concerned. Maybe not that strange when you take into account the vast frequency range of the voice, coupled with the varying fluctuations of frequencies over a given length of time.

Add to that the usual problems encountered in recording the voice, like plosives and pops, and you can see that half the time engineers are applying *Corrective* EQ, as opposed to *Creative* EQ.

And if that wasn't enough, singers then go and change the gains across their vocals and fluctuate the frequencies even further by applying vibrato to their voice whilst singing, and it is now even clearer as to why this area of EQ treatment is regarded as the '*Holy Grail*'.

Conquer this and you have found enlightenment.

Whereas drum sounds are one shot and static, in terms of their frequency ranges, the voice encompasses a whole plethora (love that word) of frequencies and at varying amplitudes over a given period. And it does not end there, oh no. You need to take into account the frequency ranges of the instruments incorporated within a mix, and compensate for that as well as trying to line up the voice in the mix.

If all the elements of a mix were static in their frequency ranges, then it would be easy. But since music is not like that, then we have the problem of finding a happy location for each and every instrument and the voice. Now, if the voice has a habit of varying it's frequencies, then it will 'invade' the spaces allocated to instruments.

We will be left with frequency overlaps, clashes, phase, etc. So, the task of 'fitting' the vocal into a mix becomes a touch complex.

Now, let's take that a stage further and introduce what really takes place in the real world, *multiple voices*.

A song will invariably have more than one vocal, irrespective of there being only one vocalist. Choruses will entail more than one vocalist, or more than one vocal line. In other words, apart from having another vocalist singing on the song, you will have a number of vocal lines from one singer, or more, that will go to form the chorus or any other part of the song. Add to that the changes in the tonal character of the way the song is sung, and you are faced with even more variables to take into account.

Ok, so the above sounds as if you are going to go through a living hell when trying to EQ vocals but fear not, it is not that bleak.

As we have covered earlier, there are ways to analyse the frequency range (*spectrum*) of any sound, be it an instrument or a voice. The methodology is the same.

The *spectrum analyser* is a great tool, so use it. But remember, your ears are the best tools available.

The trick in applying EQ to vocals is to assess what the rest of the audio is doing around the vocals.

I often enrich the frequencies around the vocals and then place the vocals in a central frequency band, smack bang in the middle of the mix.

This works great for some Dance based music, but not for R&B. In Dance music, I like the vocals to stay rigid in it's frequency home, and let the music bounce around it.

With R&B, I do the exact opposite, as the vocals are far more dynamic and flowing, so require a far broader frequency range to move in.

We also need to consider separation and layering with vocal lines, when we talk about EQ.

As we discussed earlier, EQ is a great tool to use if you need to separate instruments and vocals in a mix, but we didn't really touch on using EQ when we layer sounds, specifically vocals.

For now, I would like to briefly touch the subject of *Primary* **and** *Secondary* **EQ**.

14. Primary and Secondary EQ

I have come across a number of producers who, like myself, like to EQ the vocals *WITHOUT* the music, and then place it in the mix at the time of mixing/production. There is a vein of thought here that could be summed as 'clever'.

The idea behind this thinking is to get the vocals as clean and as dynamically strong as possible, prior to introducing it in the mix. This will then afford a clear idea as to where the vocals sit in respect to the other elements in the mix. This works quite well, and is a method with merit.

With **Primary EQ**, you EQ the sound to make sure it is clean and dynamically strong, *PRIOR* to placing it in the mix. This type of EQ does not entail colouring the sound, only making sure it is clean and neutral in terms of frequencies.

I adopt this method for additional reasons.

By only dealing with the vocal files, I can examine the details that are crucial for a good production. Most notable is '*noise*'. It is far easier to isolate and eradicate noise from the vocals if the vocals are on their own, with no other audio interfering with it. It is also much easier to isolate pops and plosives, and any anomaly, if the vocals are not being listened to in a mix of other sounds. Finally, it is easier to analyse the frequencies inherent in the vocal files, if isolated.

If you try to perform any of the above *WITHIN* a mix, then you will not be as successful or as accurate as adopting the *isolation* technique described above.

Once you have clean and dynamic vocals, the mix will build around them far more accurately. By preparing your vocals prior to any mixing, you will give yourself the luxury of easier *fault finding* and allow yourself to be more *creative*.

Let me explain what I mean by *fault finding*. You will often encounter rogue frequencies in your mix. Something just doesn't sit right. But trying to find the 'problem' could take ages and cause you nightmares. If you *KNOW* your vocals are good, then you can look elsewhere for the problem. A problem frequency on an instrument sound is always much easier to locate and fix. A problem frequency within the vocals could take ages to isolate and repair.

Once the above has been accomplished, **Secondary EQ** can be applied. This entails treating the sound with EQ to make it 'fit' in the mix and either be complimented or be complimentary.

15. Voice Examples – Double Tracking

In production, we adopt a number of tricks to enhance and invigorate the mix.

Most notable of these is *double tracking* vocals. This basically means that we take one vocal line, copy it and then treat it, and use it alongside the original vocal line. The best example of this, and the one I am going to start the walk through examples with, is using more than one copy of a vocal line to create depth, width and harmonies for the chorus.

Fig 34 is a snapshot of one vocal file, imported 3 times on 3 separate audio tracks, and treated with 3 separate EQ units..
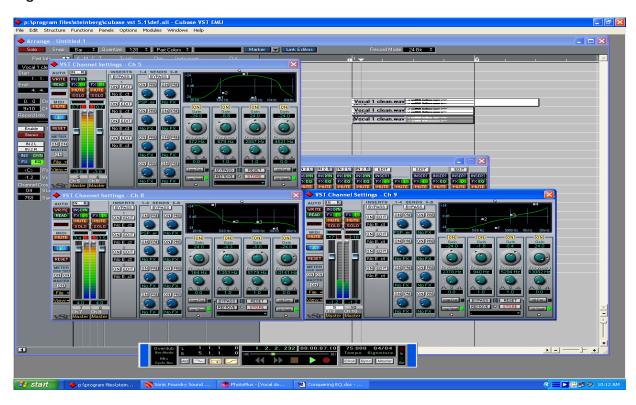
What I have done is to then EQ each one with emphasis on the 3 different frequency bands, low, mid and high.

The file will sound very similar to the original file, in fact almost identical.

Listen to the original file here (**Vocal 1 clean**).

Listen to the processed file here (**Vocal 1. 3 way EQ**).
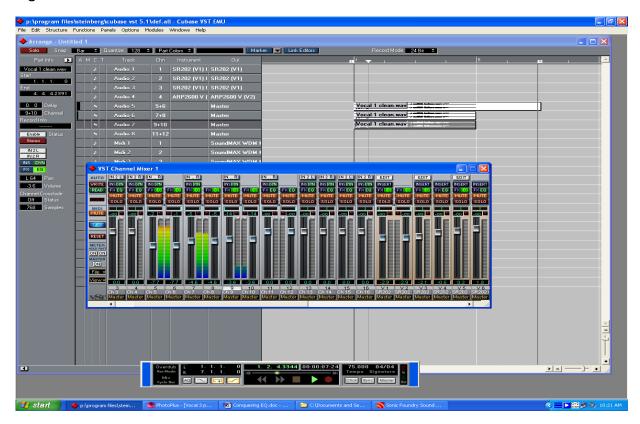
Fig 34

The reason all 3 files sound so close to the single original file is because I have shaped the separate frequencies such, that when they are all combined (or *layered*), they will look and sound the same as the single original file.

Now let us look at what happens if I pan the 3 files to selected positions in the stereo field (**Fig 35**), and more importantly, listen to the treated file.

Fig 35

Listen to the panned file here (**Vocal 1. 3 way EQ panned**).

By panning the individual files, we are able to hear the EQ treatment far better than having all the files central.

You can now compare the files and it is clear that the files, that have undergone EQ and panning, sound wider and than the lone original file.

This technique is a great way to widen those vocal tracks, and add some depth in the process.

By using the original untreated file, making copies of it and running the copies alongside the original, and applying different EQ curves, will afford us huge flexibility and variety.

You can alter the EQ curves and create further textures and pan positions.

We can take this a step further and create great effects for the vocals in our mix.

Fig 36 shows the same vocal line being treated as before, but this time we are going to use some extreme EQ curves for the copies and create interesting effects.

Fig 36

Listen to the file here (**Vocal extreme EQ**).

I have deliberately left a frequency anomaly in this audio file, just before the final vocal line, so that we can use a notch filter to remove the rogue frequency (**Fig 37 and Fig 38**).

Listen to the anomaly here (**Vocal extreme EQ anomaly**).

I have highlighted the rogue frequency so that you can easily see where it sits.

Fig 37
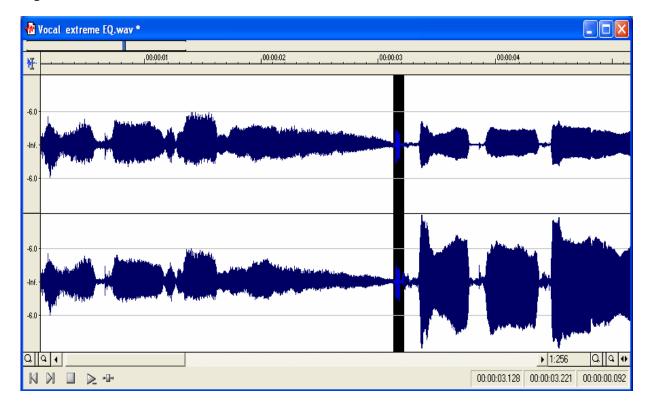
Fig 38
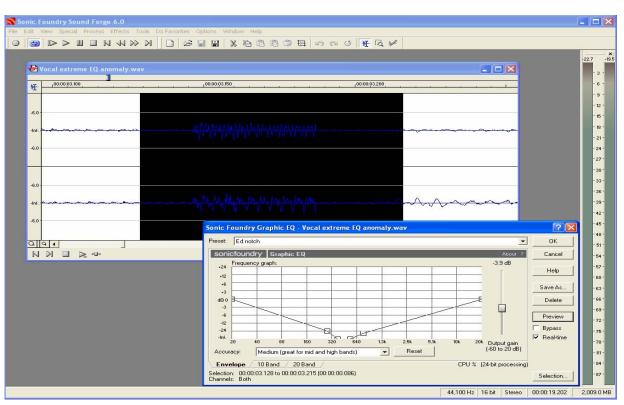
Fig 38 is a zoomed in version of the same file.

By zooming in tightly on an area, we can easily determine the nature of the problem that might lie in there. In this case we have found the anomaly.

As you can see, I am using the graphic EQ here but have shaped a notch filter to remove the anomaly at the desired frequency range.

The ears can be your best friends here. By listening to the anomaly, while adjusting the nodes on the notch filter, we can '*hear*' the changes we make.

I have now made the changes and have rendered the process.

Listen to the result here (**Vocal extreme EQ anomaly notched**).

That is what we call *Corrective* EQ.

Any EQ unit that allows you to have detailed control over a frequency range can only be a positive tool. To fully utilise a notch filter, we need to be able to home in on any frequency band and have the ability to remove or reduce the gain of that band.

Notch filters are also used for pops, clicks, plosives etc.

With vocals, I never like to give frequency ranges as I feel that the figures are so arbitrary that it would not be a good guide.
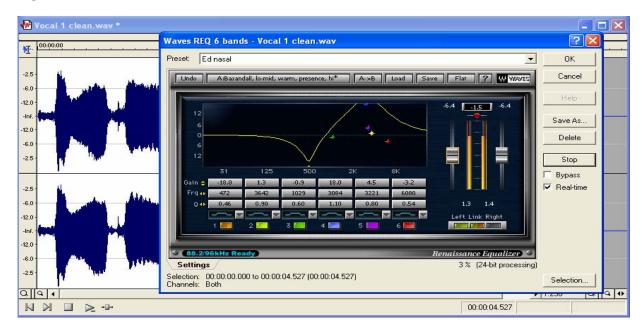
You will see graphs and list for vocals ranges, but almost all will be neglected when it comes to real studio sessions.

Use the *Spectrum Analyser* and your *ears*. They are better than any chart.

16. Voice EQ Additional Examples
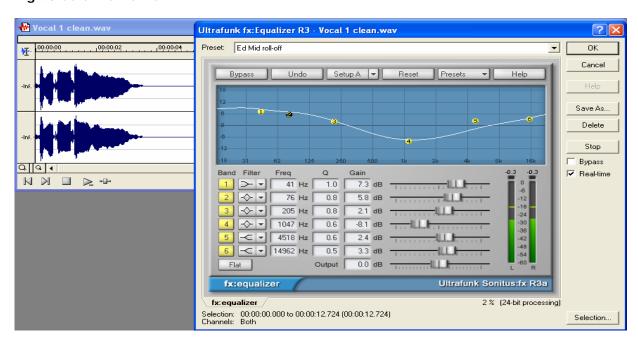
Here are a few more EQ curves for the types of effects that you hear quite often in commercial releases:

Fig 39 The Nasal Effect

Listen to the nasal effect (**Nasal effect**).

Fig 40 Soft Mid Roll-off

Listen to soft mid roll-off (**Soft mid roll-off**).

Fig 41 Warm Total Frequency Range

Listen to warm total frequency range (**Warm Total Frequency Range**)

Fig 42 Very Soft Top

Listen to very soft top (**Very Soft Top**).

If you study the examples above, you will notice that the EQ curves that cut, as opposed to boost, have a more gentle effect on the sound, and can be quite pronounced even with small cuts. The combination of cuts and boosts help to shape the sound in more detail, and to make

the overall curve look more natural, the only exception being the 'nasal effect'. For this, we need to zoom in on a select frequency band, and treat that band, as seen above.

Whenever you create an EQ curve, name it and save it.

With vocals, there are certain frequency ranges that might be worthwhile to note, just in case you need to rely on inputting values into an EQ unit, as opposed to using nodes and 'drawing' the EQ shape.

Between 3.5 kHz – 6 kHz, you will find the main 'presence' of the vocal. Boosting this will add more to the low end of the vocal line and cutting it will place the sound further away and also allow the sound to be thinner.

6.5 kHz – 12 kHz is where the 'breathiness' and sibilance of the voice resides. This is the area that can be very useful when it comes to removing anomalies or boosting the breath effect.

14 kHz – 20 kHz is the range that the crispness of the vocal resides in. This is the area that needs attention if you need to soften the delivery of the vocal, or boost if you need the more pronounced characteristics of the vocal.

The above is only a guide and will be different for different voice qualities. However, it is worth noting.

I think that will do as far as vocal is concerned.

We need to move on now, to the area that incorporates all that has been addressed here in this tutorial, the area of EQ for *Mixing Projects*.

17. Mixing EQ - Gospel

So, we have come this far and all that is left is to tidy up with a couple of examples of how to use EQ in separating tracks for mixing, and using EQ for reshaping certain elements of a mix.

By now, I expect you to be familiar with all that has been discussed in this e-book tutorial. Armed with this knowledge you should be in a position of strength, and I expect you to fully understand what I am going to show you in the following examples.

When it comes to EQ treatment within a mix, there are certain pointers that you must take on as '*Gospel*'.

1. Boosting frequencies is not encouraged unless a very specific effect is required.

We covered the subject of phase, anomalies, overlaps, and pretty much everything else associated with boosting frequencies.

If you have to boost then make sure it is in small amounts and with sensible band selection.

2. Cutting frequencies will dramatically improve your task of separating instruments and vocals.

Taking away, in this instance, amounts to giving.

We have been through this in detail. Always think ahead and do not get confined by huddling all your frequencies into a narrow range. Remember that the more range you have in your mix, the more the dynamic movement, space and depth.

No single sound should dominate. If all frequencies are correctly treated, then the listener will enjoy the mix much more, and will find something different in the tonal content every time he/she listens to it.

3. Do **NOT** EQ the final mix. Let the mastering house take care of this as any dynamic processing is destructive. Destructive means it cannot be reversed. The mastering house will not thank you if you have squeezed the dynamic range of all the frequencies into a narrow band, or tried to perform the old, highly mistaken, trick of boosting all the frequency bands in the hope of adding '*Oomph*' to the mix.

You can, of course, EQ a stereo sound, but try not to apply EQ to an entire mix.

4. You can vary the types of EQ units you intent on using on your mix, but try to use the same types of EQ, or EQ units from the same manufacturer, on the same mix. The reason for this is that EQ units, software or hardware, have their own characteristics '*colour*' (sound).

I have often used different EQ units and different types of EQ successfully on the same mix, so it is not a hard and fast rule.

5. When using EQ on individual sounds within a mix, always keep an eye on your levels, especially when boosting frequencies. Sometimes, the smallest EQ boost can take your entire mix into the *'clip'* zone (red).

6. The *Spectrum Analyser* can be a great tool, but do not live by it. Gauging frequency detail can be very useful, as we have seen already, but to rely entirely on the Analyser's readout can be a big mistake. Use your ears and the tools at your disposal, but always reference with your ears.

Just because it looks good, does not mean it sounds good.

7. Always start your EQ tasks with sounds that occupy extremes of frequency. Bass, high strings, deep kick, hi snares etc. You get the idea? Once you have sorted out the extremes, you can then start to fill up the space left in between, and once this is done, you can then EQ with detail to attain width, depth and separation.

It is a bit like painting. You paint the background on the canvas. You then add all the primary colours etc. And you finally finish off with the detail and shades.

8. The voice is always '*king*' in any mix, so pay particular attention to the tonal qualities of the vocal lines, both in terms of frequencies and levels and also in terms of the way the vocal lines are sung.

9. '*Cleanliness is next to Godliness.*' Keep your sounds clean. If you need to use corrective EQ to eliminate anomalies, then do so, but do not boost any frequencies.

Working from a clean slate will give you the best results. A poorly constructed sound file will play havoc with all the other sound files and make the task of mixing a nightmare. It only takes one bad sound file to ruin a mix.

10. If your sound files are already clean and dynamically strong and sit nicely in the mix, then you might find that they actually do not need any EQ treatment at all.

No one said '*You have to EQ'.*
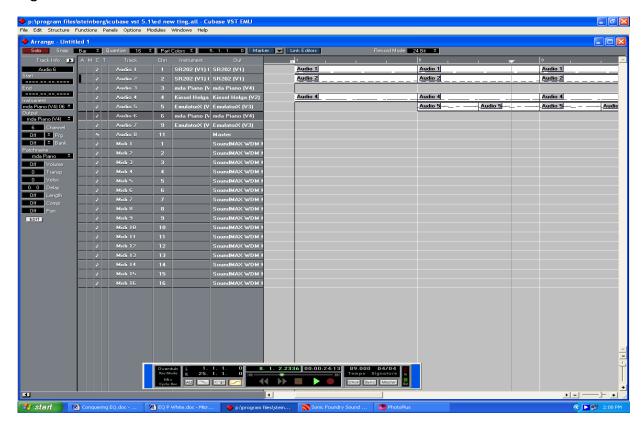
Purists will support the no EQ tradition. You have the luxury of choosing to EQ or not, but be aware that you do not have to EQ to attain a good mix.

19. Mixing EQ - Examples

Fig 43 shows a very basic mix of drums, bass and a synthesizer keyboard hook.

I have kept it simple, but used sounds that deliberately clash with each other, so that you can identify what I am doing.

Fig 43

Listen to this short mix (**Mix clean**).

You can hear that the bass is dominating the mix, and I have placed the hook right in the middle of the frequency spectrum so as to make my task of applying EQ a little harder because of the snare and hi hats. The drums sound weak and with very little spread.
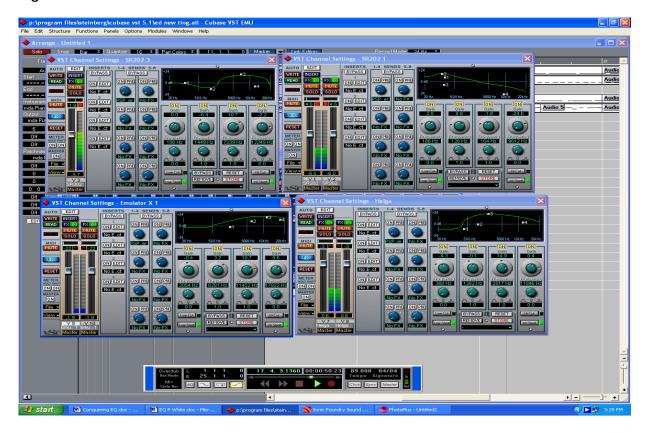
Fig 44 shows how I have shaped the EQ curves so as to give a broad overall frequency range to the whole mix.

I have maintained the hook's mid and top ranges and cut some of the low end from it.

I have boosted a narrow band for the kick and kept this space a bit vacant for the bass to occupy.

The snare and hi hats sit with the bass, but because they are already out of the bass' frequency range, they sound much clearer.

Fig 44

Listen to the final, treated mix (**Mix EQ**).

You can instantly hear the depth of the kick, the crispness of the snare, the bounce of the bass and the spread of the hook. This mix is far better. It is balanced, wide and deep and most importantly, it has separation.
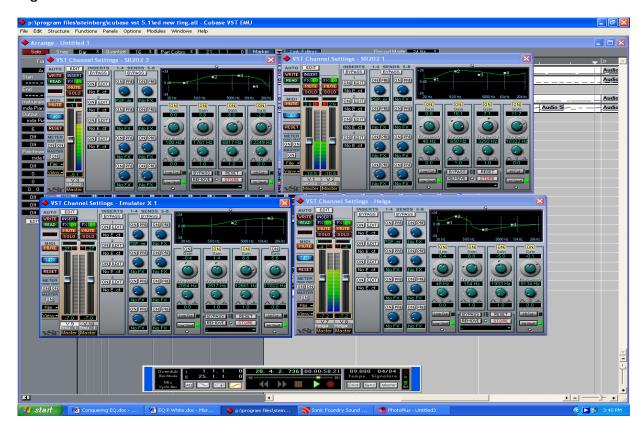
Fig 45 is a good example of a bass biased mix.

This mix is meatier with the bass having dominance and the kick being submissive and less pronounced and hard.

The snare stands out far more and the hi hats occupy a nice little niche in the mid to top-end.

This mix would not be extremely useful as the bass is too dominant, but it does show you how you can shape and shift frequency ranges to develop a totally new '*feel*'.

Fig 45

Listen to the new bass mix (**Mix with bass EQ**).

Fig 46 is our final example and it shows how to create a snare dominant mix, whilst keeping the kick deep and heavy, the hi hats mid to top, the hook spread more around the mid and the bass just lying under the drums.

Take a close look at the EQ shapes.
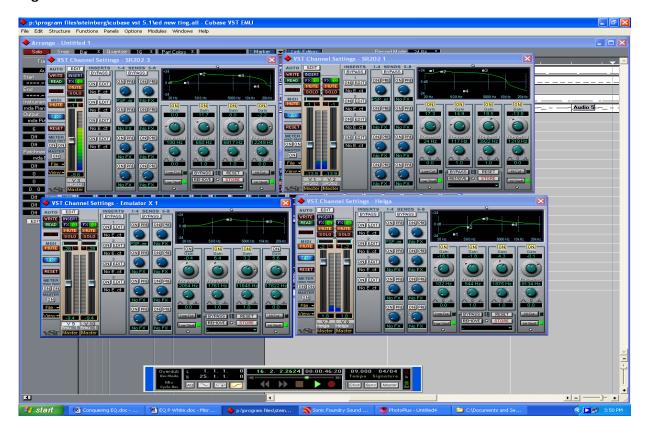
They are a combination of narrow band and spreads.

What is ve4ry clear here is that I am boosting almost all the sound files, with the exception of the bass filer, which I have rolled-off by a large amount and over a large frequency range. By doing this I have allowed the drum sounds to take over the low to mid-end, and allowed a far bigger space for the hook to feel comfortable in.

You should always strive to have fun when mixing, and one of the most enjoyable tools we producers have is the EQ.

You can create so many new textures and 'feels' by simply changing a few parameters on the EQ units.

Fig 46

Listen to the new drum dominant file here (**Mix with kick and snare**).

19. Final Word

Well, I hope that this e-book tutorial has been of help to you.

I think that the most important areas have been covered.

EQ is probably the most potent and versatile of all the tools a producer has at his/her disposal.

Use it wisely.

Always think before you jump into that 'boost' knob. Be aware of all the frequencies around the frequency you want to alter.

Cut is always better than boost, unless a specific tone is required.

Always refer to this tutorial if you get stuck.

The rest is down to you. Experiment, understand and most importantly, enjoy the experience.

Eddie Bazil (Zukan)

Samplecraze

http://www.samplecraze.com

PS. Many thanks to Jane for the vocal samples.