CONQUERING EQ

By Eddie Bazil

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Tutorial Files

Throughout this book I will refer to certain audio files – the relevant file name will be **highlighted in red** and superseded with a

The relevant file can be found in the corresponding chapter folder in the ebook download.

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

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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.

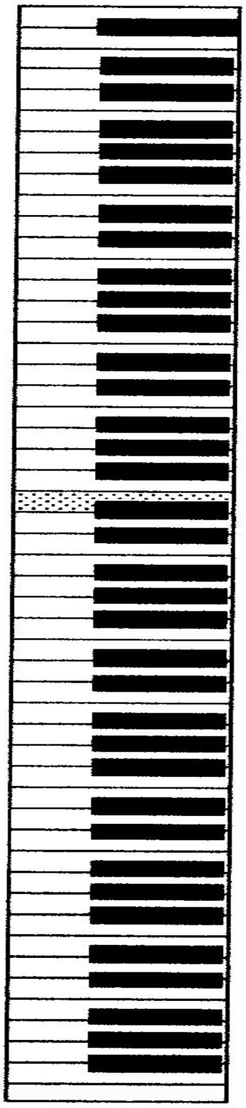
Understand this basic concept and you will be armed with the most potent weapon.

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	Midi No (sharp)	Note	Freq	Freq (sharp)
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	116.54
47		B2	123.47	
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**:

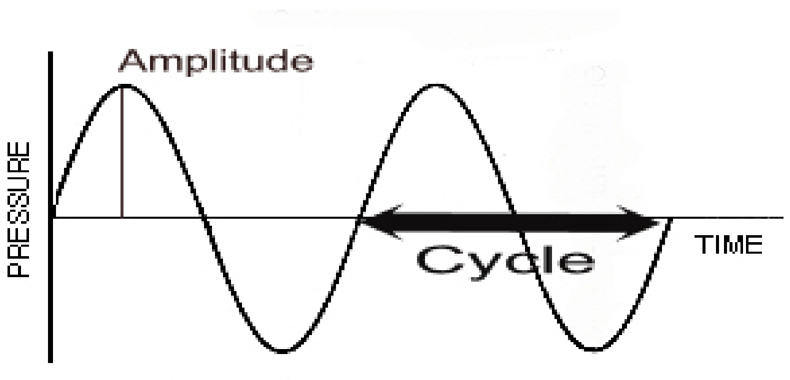
ii. AMPLITUDE

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

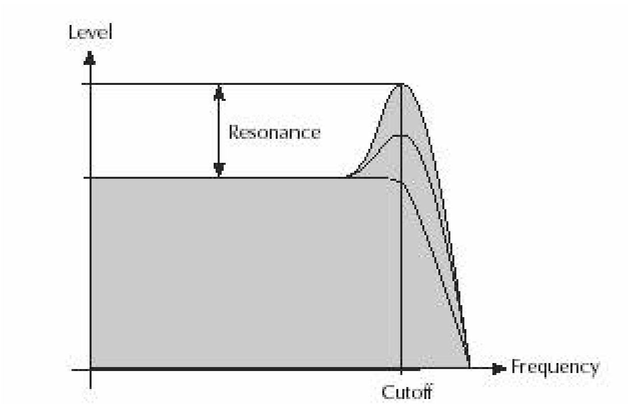
This is the point (frequency) at which the filter begins to filter (block or cut out). The filter will lower the volume of the frequencies above or below the cut-off frequency depending on the type of filter used.

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.*

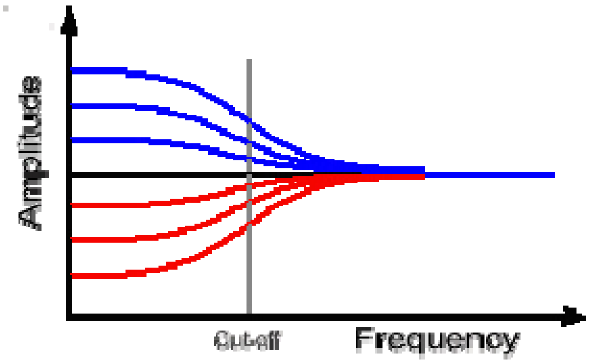
This is the simplest type of active EQ. This EQ can shape response in a number of ways: boost/cut low frequencies, boost/cut high frequencies. This is why I have included the graph to demonstrate what happens with the filters, low and high pass, in this type of EQ.

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

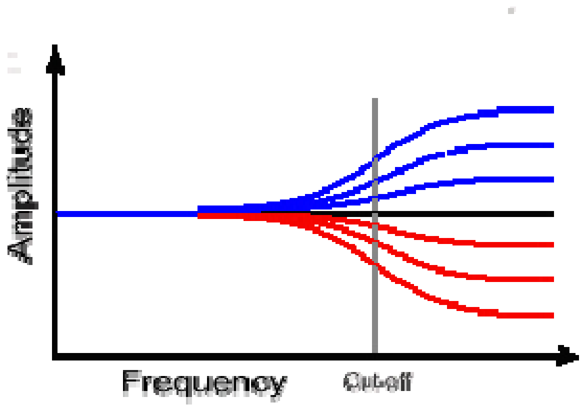
It is also common for the filter slope to be 6 dB per octave. This allows for a gentler effect. The shape is shelf like, so the boost or cut is progressive over a range. Filters do not have a no-effect at a frequency and then instantly jump and suddenly reappear at the next frequency. They have to get 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.

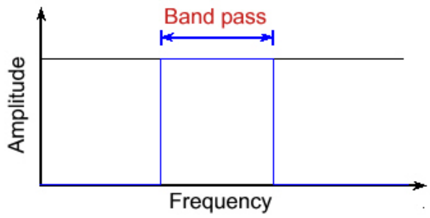
The more frequency bands you have, the more control and accuracy you have over the frequency response.

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.