Kim Lajoie's Blog Dump

Collected and assembled by Mabian

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Introduction

This document contains all articles posted (as of the date mentioned on the cover page) on the "Lifesigns from the studio" blog by Kim Lajoie, currently available at address

http://kimlajoie.wordpress.com

Even though released under permission of the blog author, this document is to be considered completely unofficial and unsupported by him.

I started reading the blog a few years ago and article after article I had the increasing feeling that the content exposed in the blog could be very useful if collected in a unique PDF file for offline reference.

So, I spent a few hours in OpenOffice to transfer the content of all articles in this document.

I quickly realized that the same information arranged this way could be interesting for everybody who likes digging into digital audio and songwriting techniques.

I then contacted Kim asking him if it was ok for him to release the document publicly; he was so kind to reply and accept shortly after and there it is.

The posts are sorted in ascending chronological order, a table of contents with links allows navigating and accessing them quickly. The PDF also has standard bookmarks and of course it's completely searchable.

Each post heading in the document is a link to the original article on the blog, so that you can easily go there to add comments, interact directly with the blog editor, download additional contents and access any other referenced material.

Thanks once again to Kim for his blog posts and for allowing free distribution of this document.

For any feedback or everything you can find and PM me (username: mabian) at the KVR forum:

http://www.kvraudio.com/forum

That's it, I wish you enjoy the document like I'm doing and have a good time with it.

I personally consider the blog one of the most awesome goldmines of suggestions, tricks and words of wisdom about the immense computer music creation world!

- Mario Bianchi aka Mabian -

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2007/06/26 - Compression and reverb

It might be helpful to look at this as a choice between "reverbing the compression" (sound->compressor->reverb) or "compressing the reverb" (sound->reverb->compressor). Reverb is usually an additive process – it adds a component (the reverberation) to the existing sound. If you add the reverb *last* (after compression), you'll be able to produce a conventional, natural sound. That's because the signal being sent to the reverb has the same (or similar) dynamic response to the final sound we'll hear in the mix. Also, the added reverb itself isn't being significantly processed, which means it will sound close to what the reverb designer intended.

By doing it the other way – "compressing the reverb", you are directly altering the dynamic rseponse of the reverb itself. This is not a common process, but may be useful for achieving special effects or unnatural ambiences. For example, smooth deep compression on a long reverb tail may lengthen the reverb tail or make it sound "deeper". More aggressive comrpession can create a very unnatural pumping effect that emphasises the reverb without washing out the original sound.

-Kim.

2007/06/26 - Ordering of EQ and Compression

The order in which you use EQ and compression will depend on the sound, what you want to do with the sound, and the rest of the mix. By using EQ first, followed by compression (sound->EQ->Compression), you are adjusting the frequency spectrum of the sound, then applying compression to the adjusted sound. You might think of this as "compressing the EQ". This is useful because the compressor will respond in a natural and predictable way because it is operating on what you hear. You can use the EQ to remove problems and shape the sound for the mix, and the compressor will respond to the "fixed" sound instead of the "raw" sound. The downside is that sometimes a compressor will adjust the percieved frequency response of a sound (usually by reducing the low end or the high end), and it's not as easy to compensate for it with pre-EQ.

Conversely, by using compression first, followed by EQ (sound->Compression->EQ), you are adjusting the dynamics of the sound, and then adjusting the frequency spectrum. You might think of this as "EQing the compression". This can be useful if your compressor is reducing the percieved high end or low end of the sound frequency spectrum – the EQ can compensate for any "funk" the compressor is adding. The downside is that the compressor is not responding to the sound you hear, which means it might not sound as natural or predictable. As an extreme example, your sound might have some significant low end rumble or resonance – if you compress before EQing, the compression will respond to the rumble or resonance even if you greatly reduce it with post-EQ.

A hybrid approach might look like EQ->compression->EQ, where the first EQ (before compression) is used to address any problems in the sound and shape it in the mix, and the second EQ (after compression) is used to add any final touches or compensate for compression "funk".

Which approach you use entirely depends on your sound and your mix. It's important to understand how it works though, so you can make an informed artistic judgement.

-Kim.

2007/06/26 - Pre-fader versus post-fader

Without going too deep into mixer topologies, the channel fader sets the gain (you might also think of it as the level or volume, though it's not quite the same thing) of the sound going into the mix bus (also called the 2-bus or the master channel). Placing effects before the fader (pre-fader) mean that those effects will "hear" the same level, no matter what the fader is set to. Placing effects after the fader (post-fader) will mean that thsoe effects will "hear" a level depending on what the fader is set to. This is particularly noticeable with effects such as compression, which respond differently depending on the level of the sound. If you set up yoru compressor pre-fader, then it will behave the same no matter what the fader is set to. On the other hand, if you set up your comrpessor post-fader, then higher fader gain will result in more compression and lower fader gain will result in less comprssion. In effect, you will use the fader to simultaneously set the audible volume of the sound in the mix AND "drive" the compression. Normally this is not such a good idea beacuse it makes it more difficult to fine-tune the mix (changing the volume changes the compression too).

Post-fader effects are typically not used often, except for sends (also called "aux sends" or "FX sends"). The "send" effectively duplicates the sound and sends one copy to the send channel (the other copy is sent through the original channel as normal). If a wet reverb is applied to the send channel, you'll have two channels making sound – the original "dry" (no reverb) channel, and the "wet" (reverb) send channel. If the send is post-fader, then the sound level that is sent to the reverb depends on the fader setting. This way, if you adjust the fader (to fine tune the mix, or perhaps automate a fade in or out) the RELATIVE level of the reverb stays the same. On the other hand, if the send it pre-fader, the absolute level of the reverb stays the same (so if you turn the fader all the way down, you'll still hear some reverb, and if you turn the fader all the way up, you'll hear less reverb relative to the original sound).

-Kim.

2007/07/17 - How to push sounds to the background

To push sounds further into the background, you don't need any magic plugins, just an understanding of psychoacoustics:

1) Less bass. Much less bass. Natural sounds that are far away will have very little bass and low mids (unless they're truly huge sounds in movies), because lower frequencies require much more power to travel. The reverse of this is the proximity effect – where sounds very close to your ear (whispering I hope!) or very close to the microphone tend to have much stronger lower frequencies. To roll off the bass, try a low-strength (1-pole or similar) high pass filter. Start low and shift it up until you no longer "feel" the sound.

2) Less treble. Less sparkle, less definition. Natural sounds that are far away will have reduced higher frequencies due to absorption by air and other materials. Distant sounds also have much less definition and clarity. Often a low-strength (1-pole of similar) low pass filter (with no resonance!) will work well.

3) Reverb, modulation. As above, distant sounds tend to have much less definition and clarity. You should do whatever's appropriate in the mix to "unfocus" the sound. Sometimes more reverb will do it. Often a very short reverb will work best. It doesn't have to be a strict room – just something to diffuse the sound. Sometimes chorus or even subtle phaser will work better. It depends on the mix – you're trying to reduce the clarity of the sound.

4) Collapse to mono. Distant sounds do not wrap around the listener's head. They're often not

“wide” (unless they’re truely huge sounds in movies). Sometimes a full mono collapse isn’t appropriate though – it depends on the sound. You might want to retain a little width in atmospheric sounds (like pads). Sometimes leaving a little width will improve the diffusion in the sound (when a full mono collapse might make it more focussed).

5) Pan centre. This works for two reasons. Firstly, sounds that are panned to the side tend to “creep up” closer to the listener. Imagine the soundstage in front of you as a semicircle – the sounds on the side can (all things being equal) actually get closer to the “front” than the sounds in the center. Also, panning centre will hide the background sounds behind other foreground typically also panned centre (such as lead vocal and snare, depending on your genre). This will make it mroe difficult for the listener to focus on the background.

6) Compose it in the background. To support the above, you should actually compose the parts as background parts. Again, this means understanding the application of psychoacoustics to composition. As listeners, we tend to focus on sounds that are:
- louder
- higher pitched
- moving quickly
- not repeating in short cycles (EDM- I’m looking at you!)
- phrased (ie. not constant)

Likewise, background parts will be the opposite:
- quieter
- lower pitched
- moving slowly
- repeating patterns
- unphrased

Likewise, background is only ever a relative measure. If your background part isn’t getting far enough in the background, it could be that you don’t have anything far enough in the foreground. Just like everything else in music – if everything is background, nothing is background.

Of course, this is all fundamental composition technique. Believe it or not, we all can learn from the classics!

-Kim.

2007/07/19 - Getting synths to fit together

Without getting too deep into ergonomics and workflow, often synth parts from the same synth (and sometimes different synths from the same company/designer) will make it easy to design sounds that blend well together. This is not only because they might share the same oscillators or filters, but also because the user interface encourages a similar approach to sound design.

On the flip side, synths with different user interfaces may encourage a different approach to sound design. Of course it's not just the interface, other factors may include different oscillator and filter algorithms, as well as different envelope curves, keyboard/velocity scaling, portamento curves, etc.

If you don't have a *precise* idea of the sound you want to design *before you design it*, you'll find yourself more influenced by the affordances of the instrument. In other words, if you think "I think a snappy bass might work here", you'll go with what the instrument guides you to – the type of sound that the instrument makes easy to design, and the type of sound that sounds good quickly on that instrument (not sure if that makes much sense – let me know if you want a better explanation). On the other hand, if you're very clear about the exact sound you want (ie, you can hear it in your head) AND you know your instruments well enough to know how to get it, then you'll "fight harder" to get what you want, but the end result will work better in a diverse mix.

If you're working on a project with several different instruments and you're finding a part isn't quite blending with the rest, try this:

1) Pull the part's volume right down to silence. Don't use the mute button – actually pull down the channel fader.

2) Listen to your mix without the part, and IMAGINE the part. This is sound design, so don't just imagine the notes or the type of sound (composition stuff – I'm assuming here you've already got that sorted). Really imagine how it sounds in the mix – frequency spectrum balance, dynamic range, depth, height (seriously!), interaction with other instruments, etc. This isn't easy, and you'll need to practise in order to get good at it.

3) SLOWLY raise the channel fader of the offending part. Stop as soon as it sounds wrong (or, you can hear the wrongness). Mentally compare the sound you're hearing with the sound you're expecting. Try to pinpoint exactly what is wrong with the sound, and what changes need to be made. Sometimes it's just one aspect of the sound, often it's a combination (which is why it's difficult to get the sound to sit in the mix if you don't know exactly what you're aiming for).

4) Fix the sound. This is where it really pays to know your tools. Sometimes it's adjusting the synth parameters. Sometimes it's different eq, compression or other effects. If the sound is very wrong and you've used a lot of channel effects (such as eq and compression), remove them. Clear the channel and start again.

If you still can't get it to work, you might need to go back to the composition. What are you trying to achieve with that part? Perhaps the rhythm isn't working well against the other parts. Perhaps you need to transpose the part up or down by an octave (or less than an octave!). Maybe your imagination has failed you and the music actually needs a different type of sound, a different instrument.

Sometimes the music is simply better off without that part. Don't try to shoe-horn in a sound just because you think it's cool – every part in the music has to support the music. Ask yourself – what is the music trying to do here? How is this part supporting it? These are difficult questions to ask, and even more difficult to answer. With practice you'll get better at it, and your music will thank you for it.

-Kim.

2007/07/19 - Mixing Synths

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-Kim.

2007/10/22 - EQing reverb

Many reverbs have some onboard EQ. Depending on your host, you may also be able to insert a separate EQ plugin before or after the reverb.

Generally, you can think about EQing reverb as three bands:

Lows: Reduce or cut the lows to cut down on mud and boom. Getting surgical here can sometimes really help clean up a mix. Increase the lows for special effects – running a kick drum through a low-heavy reverb will give you a tasty huge BOOOOOM!!!

Mids: Reducing (dipping) the mids of a reverb signal can thin it out, sometimes helping it fit in the mix. It can also help give you a very "hi-fi" sound. Boosting the mids (relatively) can increase thickness and body.

Highs: Reducing the highs can go a long way to cleaning up annoying sillibance in vocals ("s" and "t") if the reverb is catching too much of it (also think about using a de-esser on the vocal channel itself too). Reducing or cutting the highs can also make the reverb become less noticeable overall, which may sometimes help it sit in the mix better. Boosting the highs can work well when you want to emphasize the the reverb (make it noticeable) without muddying the mix.

You'll notice that I've used terms like "sit better in the mix". This is an artistic judgment you have to make (if you cut *everything* to make the reverb sit "best" in the mix, you won't have anything left!), and you'll have to make it in the context of the mix.

You'll also notice that I've given advice for reducing AND boosting different frequency areas. There's no simple advice like "Doing X will always improve your mix" (even the famous 500Hz dip!). Techniques will have certain audible *results*, but you have to decide if those results are appropriate for your mix, for your music.

-Kim.

2007/11/29 - Mixing with 2-bus processing

Mixing with 2-bus processing – for

As you already mentioned, mixing with bus processing allows you to work towards your "final sound" in one go. This is particularly useful when 2-bus processing is used for more than mastering (preparing a mix for a distribution format/medium). The best-known example is pumping compression. It's not so easy to put together a mix where pumping 2-bus compression is a feature element if you're not actually hearing it when you're mixing!

On the more extreme end, other processing tools are appropriate for inserting on the 2-bus while mixing. Personally, I've used things like buffer stutter effects, stereo width manipulation, filters, and even distortion. Mind you, these (even the stereo width manipulation) were not for mastering – they were "effects". They were usually automated, and only enabled for specific sections of a song.

On the more subtle end, I've heard of mix engineers "mixing into" a 2-bus compressor. Not necessarily to get an obvious "pumping" effect, but to gel the mix together. Apparently doing this allows the mixer to get away with using less channel compression.

Mixing with 2-bus processing – against

One of the important reason *not* to mix into a 2-bus compressor is that it can be easy to start chasing your tail in circles. It can make mixing difficult because adjusting one instrument can radically change the behaviour of the other instruments. In the simplest case, turning an instrument up will subsequently cause other instruments to drop in level when that instrument in playing. If you're not watching for it, you might later try to turn those other instruments up and wonder why the rest of the mix is dramatically rearranging itself with your every move. It can also make it difficult to compensate for (or keep) complex relationships between instruments.

Another trap is using 2-bus processing to compensate for mix problems, when a more appropriate tool would be processing on individual channels. The obvious example is EQ. If there's not enough bass (for example) in your mix, you might adjust the EQ on the 2-bus as a shortcut to adjusting the kick and bassline individually. By taking the shortcut, you're adjusting the frequency spectrum of the kick and bassline in the same way, by the same amount (when it might be better to make more tailored adjustments). You'd also be boosting the bass of every other instrument in the mix. Unintended consequences may apprear later, and you'll be scratching your head.

Yet another trap is that 2-bus processing can be confusing. Again with the EQ example – if you've boosted the bass, you might be working on a background part and be wondering why the frequency balance is skewed, when you might not have any EQ (or even different EQ) on the channel itself. With 2-bus compression, instruments will sound different when solo'd to when they're in the mix – sometimes radically so.

The other issue is the confusion between 2-bus processing and mastering. As I've mentioned before, 2-bus processing is what happens when you insert plugins on the stereo pair that goes out to your speakers. Mastering is what happens when you prepare a stereo mixdown for a distribution format or medium. If you're inexperienced, it can be easy to try to do both at once, when they really are completely different tasks (that happen to use similar tools). In short – mixing is the process of making the individual tracks work well together, and mastering is the process of making the overall sound work well in context (often next to other songs). By using mastering tools like EQ or compression on the 2-bus while mixing, it can be easy to fall into the trap of making mastering adjustments before the mix is finished. Of course, these mastering adjustments are undermined when you go back and change a mix element, which means you have to go back and change the mastering adjustments, and so on and so on.

Mixing without 2-bus processing – for

This is the "traditional" approach. Without 2-bus processing, the mix is much more controllable, and allows much more precision in making mix decisions. You hear the individual tracks as they are. The EQ and compression settings on each track actually reflect what you hear. Adjusting the levels or EQ of a single instrument doesn't magically adjust the levels or EQ of other instruments. You don't have to worry about mastering taking your focus away from getting a good mix. You can master when the mix is done without worry about later mix decisions messing up your mastering adjustments. Soloing instruments is a good way to "zoom in" on a sound, while still having the confidence that the sound won't change when the other instruments are brought in.

Mixing without 2-bus processing – against

Of course, the downside to mixing without 2-bus processing is that you can't hear the effect of any processing you might be planning to use. As I wrote earlier, this is particularly important where 2-bus processing (such as pumping compression) is a significant part of the character of the overall sound. Unless you're quite experienced, it can be difficult to balance the instruments to hit the compressor in just the right way. It's even more difficult if you can't even hear the compression because it's not plugged in! Regarding EQ, sometimes drastic EQ adjustments in mastering can reveal unintended sound elements. Mixing without such eq, it can be difficult to predict what mastering might bring out (or suppress). Arguably though, this can be mitigated somewhat by targeting your mix to be close to your target frequency spectrum. Close enough only – don't get surgical – leave that for mastering!

What I do

As a general rule, I mix without any EQ or compression on the 2-bus. Pumping 2-bus compression is not something I'm particularly fond of (for my own work). I prefer dense multilayered productions, which end up with very subtle and precarious balances between instruments. Mixing into a compressor would make this almost impossible for me. Similarly, with EQ I try to mix as close to the final spectral balance as I can in the first place without resorting to global EQ. EQ fine tuning is done during mastering, and I very rarely need to make an adjustment greater than +/-6dB.

On the other hand, I don't shy away from using 2-bus processing for "special effects". I've used filters, stereo width manipulation, buffer stutter effects, even distortion and bitcrushing. These are "special" though, and usually only engaged for specific sections of a song. They're usually automated too, for extra fun.

-Kim.

2008/05/31 - Mastering: How much should I limit?

Mastering is a process of preparing a mixdown for distribution. Often people focus on signal processing (such as EQ and compression), but it's important to remember that it also includes setting times between songs (for mastering albums or EPs), fades (usually fadeouts), and (in this day and age) encoding to a lossy format such as MP3 or AAC.

First, do no harm.

You're asking about dynamics, so I'll focus on that here.

In preparing a mixdown for distribution, you should consider dynamic range and overall level. They're not the same thing: Dynamic range is the difference between the loudest parts and the softest parts. Overall level is (in digital) the distance below 0dBfs.

If you're targeting digital distribution, then you're probably expecting your music to be added to a listener's library (such as iTunes) and played alongside other music. If you want your music to be perceived as "normal" or "professional", then you'll need your music to sound as similar as possible to the other music your listener has in his/her library.

To best achieve this, you should select two or three other songs that you think represent a similar genre to your music. The more similar (in genre, instrumentation, and overall sound), the better. The more well-known, the better. This is your reference.

You should find a way to be able to switch quickly between the references and your mixdown. Personally, I do this by starting a new project in my DAW and loading my mixdown and my references each on their own track. By working with SOLO mode on, I can switch between tracks as fast as I can press the up or down keys to switch to the current track.

Resist the temptation to immediately slap a limiter or mastering toolbox on your mixdown and turn it up to match the references. Instead, *turn the references down to match your mixdown*. Don't use meters – use your ears. Make a note of how much you turn them down.

Let's pretend for now that you have two reference tracks, and you turned them down by the same amount: 18dB.

Now you have your mixdown playing at the same overall loudness as the reference tracks. First listen to the dynamic range of your mixdown compared to the references. Are the quiet sections too quiet? Are there any loud bursts (not transients – but whole notes or sections) that are too loud? If so, you might find a compressor useful. Use it to subtly reduce the dynamic range of your mixdown. Don't use it for "colour", or to adjust transients. Typically I'll start with a fast attack, medium-long release, low ratio and medium-deep threshold. If your compressor has an RMS (or similar) sensing mode, use it. It might take quite a bit of fiddling to get this right. Remember to bypass te compressor regularly to make sure you're doing no harm. Also remember to keep the overall level of the mixdown the same. Resist the temptation to push it all louder at this stage. Typically you won't have to do much to adjust the dynamic range of your mixdown.

Once you've got the dynamic range right, it's time to look at overall level – the distance below 0dBfs. Right now your mixdown is the same loudness as your references. However, the peaks on your references are at -18dB (that's how much you turned them down), and your mixdown might be peaking around -9dB or higher. This is where the limiter comes in.

What you need to do is bring the peak level of your mixdown to -18dBfs *without changing the loudness as you hear it*.

Adjusting the peak/average level ratio this way will make it easier for you to hear when you are doing damage to your audio. The typical method of "pushing up" the level of the mix makes it difficult because you're changing two things at once – the peak/average ratio and the overall level *as you hear it*. Not only does this complicate the hearing process, but it also makes it easy to ignore audio damage because the more damage you do, the louder it gets (and as you know, humans tend to perceive louder music as "better").

Unfortunately, most limiter plugins are configured for the above behaviour, sporting an easy-to-abuse "input level" control. To get around this, you should insert two gain plugins, so your chain looks a little like this:

gain1->limiter->gain2

Set the limiter to limit at 0dBfs. When you switch it on, you should hear no effect. Then slowly increase the level of the first gain *and simultaneously reduce the level of the second gain by the same amount*[1]. Do this 1dB at a time, and listen carefully to what you're doing to your sound.

If you get down to -18dB (or whatever your references are at) without reducing the overall level of your mixdown *as you hear it* and without doing too much audible damage to your sound, then WELL DONE! YOU WIN THE GAME! Turn your speakers down, remove the second gain plugin, and render it!

If you couldn't get down to the level of your references, you need to adjust the settings on your limiter, or use different tools. Unfortunately I can't give much advice here, as it depends entirely on the kind of damage the limiter is doing to your sound, and what kind of damage you're willing to tolerate. Sometimes you might need to reduce the bass with EQ (why it's important to match the spectral/EQ response to your references before you try to match the dynamics). Sometimes you need to add some more aggressive clipping to retain the overall level as your reduce the peaks (particularly if your reference are as loud as commercial pop). Sometimes you might need to use a compressor beforehand to bring a "spiking" element into line (such as the kick drum in my track Horse Head). I've heard of some people using a series of several compressors or limiters, each reducing a small amount (personally I haven't needed to try it, it sounds far too complicated to control effectively).

If you've done well, you should have your mixdown at the same overall level as your references (as it has always been), but also with the peaks at the same level as well (-18dB in this example). When you're happy turn your speakers down, remove the second gain plugin (which will pop your mixdown up to 0dBfs), and render it!

-Kim.

[1]Unnecessary technical note: What this effectively does is move the 0dB point inside the limiter plugin! This is a cool trick to do with any other dynamically-sensitive plugin, such as compressors or distortion! Blue Cat's free gain plugins can be linked in reverse, allowing this kind of two-way gain adjustment to be done quite easily.

2008/06/27 - Sense of space in "Sunlight"

VitaminD sent me a PM asking me about the sense of space in the song Sunlight, stating that I didn't write much about spatial or panning tricks in the production diary.

I thought I'd write my response publicly here because there might be a few others here who are interested.

I didn't really use any special tools or tricks to achieve the sense of space.

Layers:
No magic here – just an understanding of foreground and background. I make sure the lead vocal, snare, kick and bass are in the foreground. Everything else goes in the background. This ensures there is a noticeable distance between the closest elements and most distant elements. Obviously this contributes a great deal to the sense of "space" in a mix.

I also made sure that the foreground elements are all quite "thick and full", and all the background elements are actually quite thin. This ensures that the background elements to not obscure each other (at least, not more than I intend), and they don't get in the way of the foreground.

Panning/spatial:
Again, no magic here. Just old-fashioned panning. Foreground elements are in the centre, background elements are more spread. If I remember correctly, I only used one mix reverb – CSR Hall - and even then it's not used much. The front-to-back distance that I wrote of above allows me to create a sense of depth without drowning the mix in reverb.

Explosion at 2:16:
The widening you hear is just an automated mid/side balance adjustment on the 2-bus. It's made more dramatic because in the preceding section I (sneakily) slowly collapse the mix to mono.

Compositionally, I do a few other things to dramatise the explosion. I take out the drums directly after the explosion, removing the listener's main rhythmic reference. This give a sudden "floating" (or even "flying") sensation, as the vocal continues forward but the "feet" are gone. In the preceding section, I also obscure the vocal (with DSP), creating in the listener a sense of incompleteness and expectation – fulfilled when the unobscured vocal is revealed again. Just before the explosion, the background instruments also go into a very short repeated loop, almost to the point of stopping altogether (notice the absence of the usual elongated chord/harmony progression at 2:08). This has an effect of reducing the scope of the listener's memory, as each logical "chunk" is much smaller. When the explosion hits, it's almost as if *the very fabric of the music is suddenly huge*.

Back wall reflection:
In the chorus there is a delayed, reverbed copy of the first few lines of the vocal. Again, nothing spectacular. It's really there to exaggerate the front-to-back depth during the chorus. I only applied it during the chorus because this sort of thing loses its effect if it's audible throughout the whole song. Doing this also emphasises the difference in depth between the verses and the choruses.

So, no magic!

-Kim.

2008/07/04 - Balancing kick and bass

A kick drum can be loose, tight, deep, percussive, hollow, tubby, thick, light, etc. Do you want it to hit you in the stomach? The chest? The face?

Likewise, a bass could be rough, smooth, deep, throaty, thick, etc. Bass often also has the potential for emphasising the higher-register sound of upper overtones. A bass could be as deep as the stomach (or even lower!), or high enough to be a low-register melodic counterpoint to the melody.

Of course, these are just examples. Kicks and basses can be anything you imagine.

My point is that you need to make a decision about what you're trying to do with the kick and the bass. They should not only complement each other, but also be appropriate for the production values of the song.

For example, if you want both the kick and the bass to be equally thick and low, the effect will be muddy and confusing if they're playing at the same time… just as if a guitar solo and lead vocal were to sound similar to each other and be playing at the same time. No matter how you EQ or compress, you will be fighting every step of the way.

The choices you make will determine the kind of processing you need to do.

Personally, I usually take one of two approaches. I might want a deep strong bass, in which case I'll make the kick weaker and more percussive. Other times I might want a higher, more melodic bass, which allows me to make the kick lower and longer. If I want a strong kick, I need to make the bass higher and/or weaker.

Which approach I choose for a song depends on what I'm trying to do with the song as a whole. For example, if the song has a strong rhythmic element or has a lot of other melodic instruments (such as guitars or keyboards) I'll usually emphasise the kick and push the bass back a bit. Alternatively, if the song is more sparse I might need the bass to provide much of the harmonic support, so the bass becomes high(er) and stronger, and the kick is lower and more subdued. If I want a heavier sound, I might want the bass to be quite low and deep, and emphasise the percussive nature of the kick (shorter and sharper).

-Kim.

2008/10/28 - High-end gear

I just want to say something about high-end gear. I'll try to keep it short.

Of course there's a difference between high-end gear and low-budget gear. Of course music made with high-end gear *generally* sounds better than music made with low-end gear. There is an undeniable correlation.

However, do not make the mistake of assuming correlation implies causality.

That means… just because good-sounding music is made with high-end gear, it doesn't mean high-end gear makes the music sound good.

High-end gear only makes music sound good because the engineers choose and operate the high-end gear. Top engineers choose high-end gear because it helps them get the sound they want. There's a point that I'm making in framing that statement – it's the engineer that gets the sound, and they get it by using the gear. The sound comes from the engineer, not from the gear.

Maybe it'd help if I put a slightly different perspective on this. Consider freeware plugins versus expensive payware. There's no doubt that someone could mix some professional-quality (whatever that means) music using nothing but freeware. So why pay for any plugins at all? Try flipping the question - *Why limit yourself to freeware?*

We choose gear not only on the sound it makes, but on other factors too. Ease of use, precision (and limitations) in controls, design and layout, compatibility, reliability, support and other factors.

When you're dabbling in a home studio, some of these factors are unimportant. We expect that if we use freeware we don't have much choice in support or ease of use. If your time is cheap, you're having fun, and you're learning, you don't mind much if it takes you ten seconds or two minutes to get the right EQ curve. You don't mind much if you upgrade your DAW (or switch between Mac and PC) and have to go searching for new plugins because your old ones aren't compatible anymore. You don't see much difference between your comrpessor having fifteen knobs with ridiculous ranges (and millions of settings that sound bad), or five of the right knobs and a carefully-tuned set of algorithms that almost never sound bad.

I'm not saying these factors go unnoticed, but that if you're starting out they're not important. On the other hand, if you're charging by the hour, these factors are *critical*. They are make-or-break. A professional mix engineer could do a mix in your studio with your freeware plugins and still turn out a decent mix, but it'd take much longer, be much more painful, and would not be as close to the creative direction.

Another example: My first real microphone was an SM57. I'd recommend it to anyone as a first mic. It's also in every recording studio too. Trent Reznor recorded the louder vocals on The Fragile using an SM57. Bjork's Vespertine was recorded using an SM58 in a hotel room. Sometimes, however, low-end gear is not the right tool for the job. I bought an LDC mic when I realised I needed more top-end clarity on my vocals than the 57 was giving me. Likewise, most studios have a mic cupboard with mics ranging from under a hundred dollars to several thousands of dollars.

In the light of those two examples, I hope the following makes sense:

High-end gear in studios is not about high-end for its own sake. It’s about not being limited to low-end gear. It’s about being able to choose the gear that helps YOU get YOUR sound.

…of course, you need to be intimately familiar with YOUR sound, and how you want to get there (which is much harder than it sounds). If you’re not, there’s no difference between high-end gear and freeware. You still suck.

One final analogy: A master concert pianist will choose an expensive grand piano to practice on, because the pianist can take advantage of the nuance and expression available on such an instrument. For the pianist, there is a world of difference between the right grand piano and a crappy school upright. The pianist lives entirely in a world where controlling the tiny nuances is the very thing that makes her a master concert pianist.

A beginner, of course, will sound like a beginner no matter what piano he plays.

-Kim.

2008/11/19 - Thoughts about coherent drum kits

The solution is equal parts creative direction and technical skill.

In other words, this is like most issues in the home studio[1] – you need to know what you want AND know how to get it.

On order to create a coherent kit, you need to know what a coherent kit sounds like. As an engineer, this needs to be much more than simply “I’ll know it when I hear it”. You need to specifically define what you want. This actually goes beyond coherence – it’s about your vision for the drum kit including what it will contribute to the mix and support the song.

As you’ve probably noticed, good sounds and right sounds are not always the same. Simply collecting a snare, kick, hats and crash that all sound “good” on their own doesn’t necessarily mean they’ll work well together and it doesn’t mean they’ll be the right choice to support the song. Don’t worry about finding good sounds and instead focus on getting the right sounds.

But how do you know what the “right” sounds are? This is where creative direction comes in. This is where you imagine the sounds (often literally hearing them in your head) before they come out of your speakers. If you do this, you’ll find that coherence is not really a problem at all.

In fact, coherence might be a red herring. When we talk about coherence we often take it to mean several sounds having similar characteristics. In the context of a drum kit, this might mean that all the drum kit elements are short and spiky, or thick and heavy, or dull and visceral, or large and fragile, etc. In fact, contrast in drum kits can also be very effective. A low long kick can be coupled with a short stark snare (eg. Massive Attack’s Teardrop). A 808 kick could be coupled with an acoustic snare (eg. Saul Williams’ DNA, or 2 Pie Island & Flue’s Little Things). An electronic kit and an acoustic kit could both be used in the same song in different sections (eg. Marilyn Mason’s This Is The New Shit).

Initial choice of samples (or other sound sources) is just as important as appropriate processing. Choice of samples is particularly important and this is where your 10GB of drum samples may be mroe a hinderance than a help. Do you really want to audition a gigabyte of kick drums every time you start a song? Likewise for snares, hats, crash cymbals, etc. You'll probably find that restricting yourself to ten or twenty good samples will actually help you improve your productivity. Don't hoard samples just in case you need one that happens to be the "perfect sample". There is no perfect sample. A sound is perfect because it's the right sound for the mix, for the song. A sound can't be perfect for the song before the song exists! Instead focus on choosing a sound that's most suitable from a collection. Be sasitfied with the sound being about 50%-80% of the sound you have in your head. Bring it into your project and bring it the final 20%-50% using processing (which may also include pitch and length adjustment). Of course, this only works if you already have a sound in your head to begin with (this is creative direction!). If you don't have this, then no amount of samples or processing will help you *because you don't know what you want*.

-Kim.

[1] Traditionally, the producer and engineer are two different people. The producer provides the creative direction and the engineer implements it. For example the producer might say the drum kit needs to be more snappy and the engineer would do that using whatever tools are available. It would be just as out of place for the producer to start talking about comrpession ratios as it would be for the engineer to start coaching the band. In the home studio, the artist is composer, performer, producer and engineer (plus many other roles too!)

2008/12/21 - Compressor attack and release times

I've put together this quick demo.

Here's what to listen for:

Fast attack, medium release
Pay attention to how the fast attack immediately clamps down on the drum transient (the initial spike). Note how the impact of the drum sounds much softer as a result. This makes the drum sound squashed.

Medium attack, medium release
Notice how with a slower attack time, some of the transients are let through by the compressor before the gain reduction begins. As a result, the very start of the drum is significantly louder than the body or tail. This makes the drum sound snappy.

Slow attack, medium release
Compare the length of the drum transient between a medium attack time and slow attack time. Notice how the "clamping down" action is slower, letting more of the transient through. As a result, the transient is more solid, and has more body.

Medium attack, fast release
Notice how, compared to the previous examples, the tail rises up much faster than before. Because the gain reduction returns to unity about halfway through the beat, the last part of the tail decays naturally. This produces a swelling sound, in contrast to the sucking/reversed sound of a medium release.

Medium attack, slow release
Here the gain reduction is not given a chance to return to unity before the next hit. What you're hearing is only the first half of the release envelope. Notice how the effect is not so dramatic. This is also because the compressor is already reducing gain when the next transient arrives – meaning

there is less room for the compressor to further reduce gain (as compared to the gain reduction being at unity just before the transient arrives). A slower release can be useful when you want to keep the dynamics controlled but in a more natural (and less dramatic) way.

Each example was processed by one instance of a notorious free compressor with no parameter readout – so don't ask me for the numbers, I don't know what they are. I tuned the compressor by ear, and so should you!

-Kim.

2009/01/01 - Using chorus, phaser or stereo imager

Chorus
Choruses create a "doubled" sound by adding a delayed copy of the sound. The delay time is very short (usually less than 40ms) so it blends with the original sound (and doesn't sound like two different sounds). Flangers are a special case of chorus effects where the delay time is usually less than 15ms. To add movement and interest, the delay time is slowly changed. This also changes the pitch of the delayed sound, which helps make the sound richer. More sophisticated chorus effects add multiple delayed copies (also called "voices") for an ever richer and smoother sound. The downside is that the sound becomes more diffuse and washed out. It can also blur the sense of pitch in the sound.

Use chorus to widen the stereo image when you want the sound bigger and more diffuse. Don't use chorus when you want to keep the sound focussed and direct.

Phaser
Phasers usually operate by a complex method of using allpass filters to cause phase shifting. You don't need to understand exactly how it works – just the sound. Phasers add movement by modifying the frequency spectrum, and changing it over time. You might think of it as a complex EQ that keeps shifting and changing.

Use a phaser to widen the stereo image when you want to sound to stay focussed and direct. Don't use a phaser if you don't want to change the tone of the sound.

Stereo Imager
This is a broad term that has been used to refer to several different techniques. Some tools widen a mono sound by delaying one side (left or right). I recommend against this technique because it causes the audio to sound like it's originating from one side (even though there's equal energy on both sides) and the tone of the sound will drastically change if your mix is collapsed to mono (or if you do any further stereo width adjustment further down the track, such as that in mastering). Some other tools adjust the stereo image by using mid/side encoding to separate the "centre" audio from the "side" audio (which represents the stereo width). They then enhance the side audio which widens the stereo image. Basic tools do this by simply raising the volume of the side audio. More sophisticated tools use EQ to let you control the stereo width at different frequency areas. The advantage of this approach is that it can sound very natural if the original audio is already stereo.

Use these tools when you have a sound that's already stereo (such as a stereo recording of an instrument or a complete mixdown) that you want to widen with a natural sound. Don't use these tools if your original audio is mono (including mono which has been processed by stereo effects).
-Kim.

2009/03/11 - How to achieve best results when mixing down to a stereo pair for mastering

A few thoughts on how to achieve best results when mixing…

Obviously, being a blog post (and not an essay!), I don't have the scope here to go into massive detail. I can, however, take a more reflective approach…

The way I see mixing, a mix engineer's job is in two parts: Knowing the tools, and using the tools.

Know thy weapon

Like any craftsperson, the mix engineer has several tools available. At a basic level almost all mixing involves adjusting the tone and dynamics of several audio tracks. Usually tone is adjusted using EQ and dynamics are adjusted using the channel fader and compression. Delving deeper, there are many variants of each and even hybrid processors that change both tone and dynamics simultaneously.

Different EQs are designed with different focusses and trade-offs. As such, different EQs *sound* different. Sometimes this is because the internal design is very unique – two EQs might sound differently even with the same front-panel settings. Sometimes, however, it is the front-panel design itself that makes the EQ sound different. The number of bands, knob ranges and discrete settings vs continuous settings will make you, the mix engineer, *think differently* when you're using it. For example, you might be dealing with a voice recording that is too muddy. Listening to it, you might decide to reduce the energy in the lower mids and boost the top. The exact frequencies you use and the amount of boost will depend on the design of the EQ you're using. Even the number of EQ bands you use might vary from EQ to EQ. To see this for yourself, try EQing a sound with a particular EQ, then take it out of the chain and use a different EQ with a different design. It's likely you'll find that, knob settings aside, you'll come up with a (slightly) different *sound*. Your mind is working differently. It's taking a slightly different problem-solving approach, dependant on the capabilities, limitations and suggestions of the tool in front of you.

The same applies to compressors and other dynamics processors. The internal design affects attack and release curves, program-dependancy and other ways in which the device responds to the sound and the sound responds to the device. The front-panel design also affects the way you, the mix engineer, thinks about how to apply compression. Do you have the full compliment of knobs (attack, release, threshold, ratio, makeup)? Are there additional knobs (eg. knee, detection mode, etc)? What about the relative size of the knobs? You might configure the compressor one way if all the knobs are the same size and lined up neatly, or another way if the threshold and ratio knobs are large and the attack and release knobs are small.

Hybrid processors can come in many flavours. At one end of the spectrum there are EQ processors that apply some small amount of saturation (which compresses the sound somewhat). At the other end of the spectrum there are compressors that apply some tonal shift in addition to gain reduction. In between is a strange world inhabited by dedicated saturators, dynamic EQs, multiband compressors, harmonic exciters, and more extreme tools like amp simulators and filters. These tools are even more diverse than regular EQs and compressors, and as such the decisions that went into their design play an even bigger part on how they interact with the audio, and how you interact with them.

For better or worse

Hopefully it should be pretty clear from the above that there's no single "best" tool in each category. Of course different tools will respond differently depending on the audio you're working with. Your choice of tool also depends greatly on YOU – does the tool help you get the sound you want? Regardless of what others say, do you find yourself fighting with it to get results? More control (ie, more knobs or faders) doesn't always translate to better control. Conversely, there's no point using a tool with limited controls if you can't get a sound you like. Your choice of tools may very well be different to someone else's – even if they work with similar music to you.

Thoughts about using the tools will have to come another time… ;-)

-Kim.

2009/03/11 - Saturation, compression and reverb

Following on from the previous post, here are some tools I use:

Saturation

Voxengo Voxformer – Great for adding "hair" to a sound. It's a very dry scratchy sound, so too much can sometimes make a sound pretty gross, but just a little bit often is enough to add some life and colour to a boring clean sound.

Magix Am-phibia – A lovely thick solid sound, especially with the two adjustment controls under the advanced panel. Coupled with a VERY gentle compressor and some VERY interesting EQ and exciter options, this is simply a great tool for locking down a sound.

Magix Am-track – The tape saturation module here is great for smoothing out a sound. I tend to use it more like a coloured limiter that behaves in interesting ways when pushed hard.

Occasionally I'll overdrive other plugins (such as Stilwell Oligarc or Audio Damage Dr Device and Rough Rider) for a saturation effect, but this is on more of a "most songs" basis, rather than "most tracks".

Compression

Magix Am-track – This is my primary compressor. The modern mode is great for basic dynamics control. It does exactly what it's told to, with the exception that it never sounds nasty. I don't know how they did it, but somehow it never sounds nasty – even at extreme settings. Sometimes it'll sound *inappropriate*, but never nasty. The vintage mode is particularly good for adding some vibe to drums and percussion – it tends to shape the envelope more than control dynamics.

Audio Damage Rough Rider – For when I need something extreme, this is what I reach for. It pumps and breathes easier than almost anything else I've tried. While Am-track never sounds nasty, Rough Rider almost never sounds *nice*. It doesn't even have a neutral default setting – the plugin starts smashing the audio as soon as it's plugged in. The top-end roll off is good too, it often makes the audio sound more solid and compact. When I'm applying this kind of compression, a little colouring doesn't really bother me. If it's too much, I'll usually boost the top end going into the compressor. While the frequency response is flatter that way, it's still far from neutral and the built-in clipper does interesting things to the boosted top end before it's rolled off…

Occasionally I'll use other compressors, such as those Voxengo Voxformer (for convenience) and Magix Am-phibia (for something special).

Reverb

IK Multimedia CSR – What can I say? I love it. Feels just like it should. The hall and plate are my favourite at the moment. I've got several of my own custom presets that I use about 90% of the time (often tweaked slightly for the song). If I'm doing something more conventional (such as Erin Shay's A Day Too Long), I'll start with one of the factory presets and adjust it for the song. At the moment I'm rolling all the sibilance off the top, creating a darker sound that sits easily in the background without drawing too much attention to it.

-Kim.

2009/03/15 - On stereo widening

I have some thoughts on stereo widening.

There are several different processes that are commonly described as "stereo widening". These include mid/side rebalancing, haas-length (less than 30ms) ping pong delays, and even inverting the polarity of one side.

Without going into too much detail, I consider these processes to be useful mainly as special effects. Of all stereo widening processes, mid/side rebalancing is the most natural-sounding (and mono-compatible). Even still, there should hardly be any need to use it. Pan and stereo width is one of the least-effective tools in composition and production. This is because most people don't actually listen in stereo! I've lost count of the number of houses I've seen where two "stereo" speakers are actually pointing in different directions, or the number times I see people listening to only one earbud from their iPod (most commonly when introducing people to new music!).

Stereo widening also shouldn't be necessary to separate sounds in the mix. Effective front-back depth can be achieved when mixing using the regular tools for adjusting tone (EQ), dynamics (compression) and ambience (reverb). If you're composing, front-back depth can also be achieved by manipulating register, motion and familiarity. I've written more about this here.

Using stereo wideners instead of panning is similar to using harmonic enhancers instead of EQ.

Who is your target audience?

-Kim.

2009/04/16 - How do you process your bass?

It's a funny question, that.

The first response that comes to my mind is: "Use your ears and figure it out yourself!". Yeah. Ha ha. Like I'm going to reach through the Internet and magically train your ears for you.

But then I realise that people have all these astonishing tools available to them in software form (EQs, compressors, saturators, exciters, enhancers, various filters and modulation effects). And they all come in different flavours. And combinations. And each vendor is spruking their own brand of magic mojo. And everyone on the internet is talking about side-chained compression. And Mid/Side processing. And multiband compression. And layering sine waves. I guess it can be a bit daunting if you don't really know how to navigate this mess.

And in many modern music styles, bass is pretty important.

But there's still the problem of context. How you process your bass depends entirely on what it sounds like raw, and what you want it to sound like. Processing is just a way to get from one sound to another. Giving you my personal settings won't work for you because your raw sound isn't my raw sound, and my desired sound isn't your desired sound (we're working in different styles of music, with different mixes and different creative directions). Add to that the fact that we're using different toolsets, in different studios.

So perhaps my arrogance is actually a way of avoiding the root problem: You want help, but I can't hold your hand.

Maybe instead we can talk about some of the different tools you might want to use, and how using them might help you get from your raw sound to your desired sound.

-Kim.

2009/04/16 - Processing Bass: EQ

The most important tool for processing bass is EQ. EQ allows you to emphasise and de-emphasise different frequency ranges. Pushed to extremes, it can be used to change the "voicing" of the bass – the placement of where it sits in the mix, and how the tone changes for each note.

Before you reach for EQ, though, you need to know your monitoring environment. I won't go into too much detail here (maybe another time), but your monitoring environment includes factors such as your speakers, your sitting position, the size and dimensions of your room, the furniture in the room, etc. Room acoustics are a complex and subtle thing, and you need to know your own monitoring environment well. That means listening to a lot of music!

EQing instruments is difficult if you don't know your monitoring environment. This is especially true when working on bass, because in most rooms the bass response is more uneven than any other frequency range. Compounding the issue is the fact that, mix-wise, there's very little "space" down there – it can get quite cramped.

Assuming you're ready to EQ, the approach you should be aiming to take is that of listening to your raw sound *and imagining in your head what the end result will sound like*. If you can do this, you won't waste much time "playing" with your tools. Obviously this is something that comes with experience (which is why it's important to make as much music as possible!). In the mean time, what do you do?

There are numerous "EQ guides" floating around on the internet, with various EQ ranges (or worse – specific frequencies), accompanied by descriptions of what happens when you boost or dip each frequency. While they're interesting, they're just opinion. I suggest you get your hands dirty and and explore the frequency spectrum yourself, with YOUR ears, with YOUR tools, in YOUR studio.

Grab an EQ, and explore boosting and cutting at various frequencies on your own sounds. Start to train your ears on what each frequency range sounds like. As important as the end result is, it's equally important to train yourself about *what your raw sound is like*. For example, if dipping at 100Hz brings you closer to the sound you desire, bypass the EQ and listen to the raw sound. Reflect that this is what too much 100Hz sounds like (hence why you're dipping there). Similarly, for example, if boosting at 100Hz brings you closer to the sound you desire, bypass the EQ and reflect that this is what not enough 100Hz sounds like. Get to know your sound.

Keep in mind at all times that EQ is *relative*. It's never one-size-fits-all. Not even close. The EQ settings you use depend entirely on what your raw sound is, and what your desired sound is.

-Kim.

2009/04/16 - Processing Bass: Compression

I've lost count of the number of times I've seen this exchange:

"hwo do i maek my bass PHATTNESS pls halp"

"Use a compressor"

I can't decide what's funnier – the blatent (and sometimes obviously deliberate) spelling errors of the question, or the starkness of the response.

While not as powerful as EQ, compression can be useful in turning your raw sound into your desired sound. Traditionally, the role of compression is to reduce the dynamic range of the audio by turning down the loud parts and turning up the soft parts. If you're working with a recordng of a musician performing on electric bass, you might use compression as a corrective tool where some notes were played stronger or weaker than others.

For electronic genres, however, the note-to-note levels aren't as much of a problem because the MIDI data (the "velocity" of the notes") can be manipulated directly. In fact, it's fairly common for synth bass sounds to be set up so that every note is the same level, regardless of how hard or soft you might play the keyboard. For electronic music, differences in levels can come from filters or EQs interacting with different notes in a bassline. If you're using some fairly extreme EQ and/or filtering then you might find that some notes are reinforced (making them louder) and other notes are quieter.

Sometimes it's hard to tell, though, because your monitoring environment might also be reinforcing some notes over others (in this way, the room acts as another layer of EQ or filtering). If you think your bassline is changing levels from one note to another, take a look at your level meters. If the level meters are *visually* correlating with what you're hearing, then compression can be used to even it out. If the levels are constant, then the problem is with your monitoring environment and compression won't help.

If you're using compression to even out changes in level caused by EQ or filter, it's best to place the compresser *after* the EQ or filtering. That way you'll have the most control, and the compressor will respond to what you're hearing. If you use EQ or filtering after the compressor, then that might introduce more level differences that the compressor can't fix.

When setting up the compressor, you'll want to start with a fast attack, medium release and high ratio. Set the threshold so that the quietest notes are just reaching the threshold – the other notes should be over the threshold, thus triggering the compressor. Best to choose a hard knee instead of a soft knee. If you can hear the level of the bass dropping after loud peaks, try reducing the release time (this will make the compressor react quicker to drops in level). If the bass sounds like it's distorting, increase the release time (this will make the compressor a little smoother). If you have some extreme processing and you want some notes to stand out (filter sweeps, for example), reduce the ratio (this will make the louder notes punch through a bit more).

-Kim.

2009/04/20 - Processing Bass: Character and body

Watch your levels!

While EQing your bass, another thing to keep in mind is equal loudness contours. Put simply, we (as humans) are more sensitive to upper-mids (1kHz-5kHz) than to lower mids (100Hz-1kHz). We're least sensitive to the extremes at each end (less than 100Hz and greater than 5kHz). When EQing bass, this means that the lower the frequency, the more level you need. The more level you need, the more you swamp the rest of the mix and the more you need to turn *everything* down to compensate. It can also cause problems if you're using compression or limiting on the 2-bus or in mastering.

Similarly, problems can occur further up the spectrum. We are more sensitive to lower mids than bass. In practice, this means that you don't actually need much level before it *sounds* like too much. For example, you might find that you can boost a lot of 75Hz before it gets obnoxious, but you can only boost a little at 250Hz before it gets obnoxious. This is also why it usually sounds "better" to dip the lower mids than the bass. The thing to keep in mind, however, is that the lower midrange is where the *character* of the bass is, whereas the bass is where the *body* is (and the upper mids is where the *articulation* usually is).

I can hear the cries – "I WANT CHARACTER **AND** BODY IN MY BASS!!!"

Yeah. Great. Good luck with that. It's called the volume fader.

Instead, think about character and body as being a balance. The more character you have, the less body you need to have in order to compensate. Likewise, the more body you have, the less character you need to have. Don't cry yet. You actually don't need much character for your bass to be "characterful", and you don't actually need much body for your bass to be "thick and deep". In the context of how it will be percieved by listeners, it's more a subtle shift of emphasis than a reshaping of the sound.

2009/04/21 - Processing Bass: Saturation

So, we've addressed two important processing tools available to a mix engineer – EQ and compression. Next up is one of my favourites – saturation. How can saturation be useful for bass?

Saturation can do a number of things simultaneously – it can reduce the headroom requirements of the track, it can make the bass more audible on smaller speakers (and more powerful on larger speakers), and it can help it sit more consistently in the mix.

Saturation reduces the headroom requirements of a track in a similar way to a limiter (or compressor with high ratio) – by making sure signal cannot exceed a certain level. This "chops the tops off" the loudest notes, bringing them down to the level of the other notes. Unlike limiting or compression, however, saturation doesn't do this by actually reducing the volume. Instead, using the saturation the loudest notes are distorted a little. The net effect is that pure volume is transformed into noise. Another way of looking at it is that lower-frequency energy is turned into higher-frequency energy.

What does this actually sound like?

It should sound like the bass is at the same level, but instead of stronger notes getting *louder*, they get *noisier*. Applying saturation with a deeper threshold (so that most notes are saturated, not just the loudest ones) makes the whole bassline sound noisier. What's actually happening is that upper harmonics are being generated. Because the bassline is monophonic (single notes at a time – not chords), the harmonics being generated are related to the pitch of the bass. Because these harmonics are at a higher frequency, they're audible even on smaller speakers that can't reproduce the lowest bass sounds. This means that your bassline will still be audible without requiring large speakers. You don't even need much saturation for this to occur. On larger speakers, the upper harmonics reinforce the bass, making it sound more powerful.

As for actual selection of tools and settings, you'll have to experiment. There are a wide variety of different methods of saturating a sound, and there's a wide variety of different sounds available. Aside from some kind of "drive" control, saturation processors don't have standardised controls in the same way that EQs and compressors do. Just remember that less is usually more – you don't need to push the bassline into full distortion to achieve a useful saturation effect.

-Kim.

2009/04/23 - Processing Bass: Layering

Strictly speaking, layering is not really a method for *processing*, but it's a common approach to take when designing a bass sound. Layering is an *additive* approach to designing a sound, because you're building it by adding different elements together. By contrast, a *subtractive* approach (such as subtractive synthesis) works by starting with a big sound and taking away the parts you don't need (for example, by filtering). In the real world, you'll probably find yourself combining thw two approaches.

Recall the discussion about *character* and *body*. Sometimes you might find that your main bass sound has a satisfying body (energy below 100Hz), but not much character (above 100Hz). Other times, you might find that you like a bass sound that has a lot of character, but not much body. Using layering, you can build a composite bass sound that has right body and character for the song you're working on.

The trick to making this work is to stay focussed (in your mind) about what you're trying to achieve. Otherwise it's too easy to create an indistinct mess of sound.

For example, you might have a deep filtered synth bass that sits perfectly at the bottom of your mix, but loses power when the rest of the mix gets busy and doesn't "cut through". You might try to make the bass brighter by raising the lowpass filter or using saturation, but then find that the sonic signature of the bass changes too much and you lose the characteristics that you like about it. Rather than trying to make the bass sound brighter, think about layering a second element so that the original sound stays at the bottom of the mix but the added layer adds some more character in the lower mids. You wouldn't need much – the added layer can be effective even if it's quieter than the original layer.

Alternatively, you might have a bass with a lot of character in the lower mids but find that it's not adequately covering the bottom of the mix. You might also find that the level of the bass below 100Hz varies quite a lot (especially if the bassline covers a wide range of notes). Boosting the bass might make the level even more uneven, and reducing the note range of the bass would probably compromise your bassline. Rather than trying to add more of what isn't working down there (or destroying your sound with a multiband compressor), consider adding a new layer to cover the bottom of the mix. That way you can focus the original layer on the range where it's strongest (the lower mids, or wherever the *character* is). If the bassline has too wide a range, you might even simplify it for the lower layer, so it is more focussed and sits better under the mix.

-Kim.

2009/04/26 - Spectrum Analysers

Are spectrum analysers a useful tool or a distracting diversion?

It's been written in many words and at many places that spectrum analysers are a Good Thing(tm). They let you see what you can't hear, help you overcome deficiencies in your monitoring environment, and help pinpoint problem areas.

Having said that, I don't use spectrum analysers, and I've advised others not to use them. The justification I usually provide is that it's better to mix with your ears instead of your eyes, and even if you're not very experienced, it's better to train your ears as much as you can instead of learning to rely on visual aids. After all, this is *music,* right? *Audio?* Your listeners aren't listening with their eyes…

But am I right?

Does using frequency analysers really result in poorer mixes? Do they really slow down ear training?

This is my opinion, but it's an only an opinion. A good friend once told me "An opinion is what you have when you don't have all the facts. When you have all the facts, you don't need an opinion." I'll admit it: I don't have the facts to support these assertions, and I don't think I ever will. All I have is based on my own style of working, and my own beliefs and values.

I still maintain that spectrum analysers are to be avoided, but I don't have any supporting evidence to show you. You have to try them out and make up your own mind.

-Kim.

2009/04/28 - Downwards or upwards?

Every now and then, people ask about upwards compression or upwards expansion. Often when this is raised there is some confusion about what "upwards" means, and how it might be different to regular compression or expansion. This confusion is sometimes exacerbated by a misunderstanding about how compressors work (as distinct from what they're used for).

Essentially, "regular" compressors and expanders both work by *reducing* gain – that is, lowering the volume. Compressors do it by reducing gain when the volume level of the audio goes *above* a certain amount, and expanders do it be reducing gain when the volume level goes *below* a certain amount.

Limiters and gates are extreme versions of the above. A limiter is an extreme version of a compressor – reducing the gain so much that the output level never excedes a certain amount. A gate is an extreme version of an expander – reducing the gain so much that the output level is *silent*.

Because compressors and expanders work by reducing gain, they can be referred to as operating *downwards*. You might refer to a regular compressor as a *downwards compressor* and refer to a regular expander as a *downwards expander*.

(Sometimes this is a source of confusion because compressors are often used to make sounds *louder* – because the loudest parts of the sound art turned down, the whole track can be turned up without distorting.)

In contrast, some compressors and expanders can operate *upwards*. They work by *increasing* gain. Upwards compressors work by increasing gain when the volume level of the audio goes *below* a certain amount. Upwards expaders work by increasing gain when the volume level goes *above* a certain amount.

Upwards compressors and expanders aren't too common – possibly because they can be unstable and produce *really* loud levels! Still, sometimes these terms arise and it's good to understand their meaning. In essence:

Upwards compression – reduces dynamic range by making quiet sounds louder
Downwards compression – reduces dynamic range by making loud sounds quieter
Upwards expansion – increases dynamic range by making loud sounds louder
Downwards expansion – increases dynamic range by making quiet sounds quieter

-Kim.

2009/04/30 - Vocal doubling

Sometimes people talk about making a lead vocal sound thicker by "doubling" it – copying the track and applying some subtle effect to the copy (such as delay or pitch shift), and then sometimes panning the two tracks opposite each other.

Personally I never use such doubling tricks – if I want to emphasise the vocals in a particular section I'll either add backing vocals (usually harmony, but occasionally unison), or copy the vocal to a new track and add a longer delay – to add depth rather than density.

Personally, most doubling tricks just sound like gimmicks to me. Sometimes if I really want a unison double and the singer isn't around anymore I'll use some of the other takes (with editing/correction if necessary). However, if you really want a unison double AND your singer isn't around anymore AND you only got one take (ie, you can't use the additional takes as doubles – which I recommend doing if you can) then your options are limited. Whatever you do it'll still sound like a single take, a single performance.

-Kim.

2009/05/01 - Vocal processing

My usual vocal processing chain consists of several stages: Gate, EQ, Compression, De-Essing, and reverb (as a send).

Gate

This is first in the chain so the gate has the full dynamic range of the original audio. The more natural the dynamic range available to the gate, but easier it is to set the threshold and timing for a natural sound.

EQ

Generally I prefer to use EQ before compression. This is so I can get the tone I want for the mix before I adjust the dynamic range. It also makes it easy to highpass the audio so the compressor doesn't respond to low-frequency audio (such as rumble) that isn't going to make it to the mix anyway. I've written more about the order of EQ and compression here.

Compression

I choose to apply compression to the final tone of the sound, rather than adjust the tone afterwards. This helps the compressor react smoothly and naturally to the sound we hear, rather than responding to sound that is going to have its frequency balance changed afterwards.

Saturation

I don't often use saturation of vocals. When I do though, it's just after compression, and I use it similar to a limiter - to catch the few peaks that are too loud even after compression. Usually I set it up so that loud sustained notes are saturated, making them *sound* loud without overpowering the mix, but most notes are left clean (not saturated).

De-essing

This is interesting. I've found that I get the best results by applying the de-esser after EQ and Compression. I find that the way I use EQ tends to enhance sibilance (tonal tilt toward high frequencies and high-ratio compression). Using a de-esser earlier in the chain sometimes means that the later EQ and compression counteract the effect of the de-esser. This forces me to apply more de-essing, which ends up sounding (more) unnatural (at extremes, it can "pump" a bit – but not in a good way!). By de-essing after EQ and compression, only a silght amount of de-essing is needed.

Reverb

Reverb can be quite sensitive – responding to both tone and dynamics. In the kind of dense productions I usually do, it's best to feed the reverb as consistent a signal as possible, to avoid widely-warying levels of ambience or a build up of mud. The purpose of reverb here is to add ambience and air to the vocal sound – not to be heard as an effect separate to the vocal. To that end it's important that the reverb responds to what we hear (similar to the compressor). I'll often lowpass the reverb to keep it sounding lush and avoid it "catching" any sibilance.

-Kim.

2009/05/03 - EQ and "frequencies"

Regarding EQ and "frequencies", the important thing to remember is that EQ is relative. That means that the EQ you apply depends entirely on the sound you're working with. For example, there's a lot of character in vocals at around 2.5kHz – but how much you scoop or boost will depend on the sound you're starting with, and the mix you're working with. You might also find that 1.5kHz or 3.5kHz is more effective for what you're doing.

I might cop some flack here, but there's no substitute for using (and training) your ears. Grab an EQ and experiment on your sound. try boosting or scooping at different frequencies (one band at a time!), and listen to how it changes the quality of the sound and how it changes the way it fits in the mix.

-Kim.

2009/05/04 - Compressing kick drums

Just a quickie to say that the *choice* of compressor you use for kick drums is more important than the actual settings you use. Kicks are a fickle beast – a few ms or a few db can make a huge difference in how the sound is perceived. This is also where a lot of compressors differ – in how they handle strong low frequencies and how they respond to rapidly-changing envelopes.

If you're not getting the sound you want from your kick drum, don't spend too much time adjusting the compressor. Instead try a different compressor. This is why it's handy to have a few different compressors on hand – preferably different styles. Sometimes an agressive compressor is the right thing to pump the volume, sometimes a gentle compressor is better for shaping the attack and overdriving the tail. Often a compressor won't work no matter what settings you use, and another one will work within a few seconds of tweaking.

Also – don’t assume that the compressor that worked for a kick worked last time will be the right one next time. Sure, it might be worthwhile to start with it, but keep in mind that kicks (and the mixes they’re in) are just as variable as the compressors themselves.

-Kim.

2009/05/05 - High pass filters in mixing

It’s commonly suggested that it is “correct” to apply a high pass filter to every track except the kick drum and bass. Doing this does two things:

1. It keeps the low frequencies in the mix for the kick and bass, so that they sound as pure and focussed as possible. This helps in making the low frequencies as loud and clear.
2. It reduces unwanted noise from the other tracks, which makes them more focussed. This helps in making the overall mix sound clearer and cleaner, and reduces the headroom required by this unwanted noise.

Like all mix techniques, however, there is a degree of balance and creativity required. Too little filtering may make your mix muddy and messy. Too much, though, will make your mix thin, light and empty. As producer and mix engineer, you must make a judgement as to the appropriate amount. Your approach will vary from track to track, and even from mix to mix. You will know the appropriate amount through experience and critical listening.

There are even some situations where the best approach is to avoid high pass filtering altogether. You might want to mix like this when you have a song that needs to be sparse and natural – usually with recordings of acoustic instruments and voices. I took this approach with my mix of A Day Too Long.

For other, more modern mixes, I approach each track individually. I gradually raise the high pass frequency until I hear it take away too much body. It’s often worth trying different filter slopes if they’re available. Often I’ll find that a straight high pass filter is too harsh (read: steep/extreme) so I’ll use it in combination with a low shelf EQ. I might set the high pass filter quite low – usually below 200Hz – and use the low shelf to reduce the body (lower mids) without drastically changing the character of the sound.

Even though I don’t use frequency analysers, others find them useful. Yellowfever, for example, suggests using a frequency analyser to visually find the frequency where a high pass filter wouldn’t affect the tone of the sound.

-Kim.

2009/05/06 - Processing snare (in a drum kit)

Those who come from the world of electronic music and samples may experience some shock when mixing a real, live drum kit!

Traditionally, sampled drums have been mixed with one sound per channel. There might be a kick channel, a snare channel, a hihat channel, and perhaps more channels for cymbals, toms and auxillery percussion. If you want the snare to sound a specific way, it is (relatively) simple to process the snare channel – adjusting the tone and dynamics of the sound – to transform it from its original form into the desired form.

When mixing a live drum kit, however, this is not such a simple process. If you come from an electronic or otherwise sampling background, you will probably be tempted to solo the snare channel. From there will will realise that the channel is much "dirtier" than you might be used to – the sound is very dry and raw, and there's bleed from the rest of the drumkit! You might battle for some time with gates, EQs, compressors, and perhaps even transient shapers. When you are satisfied with your sound (or get close enough without going insane), you'll unsolo the snare to hear it in context…

And be shocked that it sounds nothing like what you thought it would!

And soon enough, you'll discover the second difference between mixing samples and mixing a live kit – more bleed! Specifically, the snare drum will be picked up by several channels. This will at least be the "snare" channel (the one you meticulously processed) and the overheads. The snare may also be coming through the tom channels, a hihat channel if you have one, and even the kick channel!

Zealously gating the bleed out of all the other tracks will ultimately produce something that doesn't sound much like a live drum kit. Sorry – you can't perfectly control every aspect of the sound!

It might be more appropriate to approach the drum kit as a single instrument instead of a collection of individual sounds. Listen to the whole drum kit and focus on the snare. That sound is coming from at least three channels – the snare channel and two overheads. Solo each channel individually and listen to how the channel contributes to the overall snare sound. When you're imagining your desired snare sound and thinking about how to transform what you're hearing into what you're imagining, consider that you might have to make changes to more than one channel to achieve it. Also consider than no matter how much you may process a single channel, you will only be changing one component of the sound.

Finally, keep in mind that the more processing you apply, the less natural your sound becomes. Presumably, you're using a live drum kit because you want the sound of a live drum kit in your song. If you process it so much that it sounds like a bunch of samples, you negate the main reason for using a live kit in the first place!

-Kim.

2009/05/07 - Approaching mixing

Often there's quite some discussion about understanding individual techniques or dealing with particular sounds in mixing – how to use compression, how to process drums, how to saturate bass, etc.

While an understanding of techniques is essential, it's just as important to see the bigger picture – how to fit it all together, how to approach a mix.

The approach to take to mixing can vary greatly depending on style, taste and intent of the mix engineer and producer (with respect to the song itself!). It might also be affected by factors such as the available tools or project timelines.

The approach I use is to begin with the foreground sounds first – to start with the most important elements of the mix. This is usually the lead vocals and the drums. These are the principals, the centrepieces of modern popular music. The vocal provides the melody and emotion and the drums provide the rhythm and the groove. Once those large pieces are in place, I'll work on the supporting elements – bass, background vocals, auxillery percussion, keyboards, synths, guitars. After that I add the final spice – background sounds, edits, automation, special effects.

I sometimes liken this to a process of fitting rocks in a jar. If the rocks are all different sizes, it's best to start by placing the largest rocks in the jar, followed by the medium-sized rocks, and lastly the pebbles. You can imagine getting into trouble if you tried to start by adding the pebbles first and waiting for the the end before adding the largest rocks. You might have quite some rearranging to do! The same applies to mixing.

-Kim.

2009/05/08 - Overcoming "loopitis"

It seems that many computer music composers have a problem with developing songs[1]. There is a tendency to create elaborate multilayered loops of two or four or eight bars, but a resistence to being able to put together a whole song.

At risk of oversimplification, this is caused by *vertical* thinking, where a section of music is developed by adding additional layers (and also getting lost in mixing – balancing and processing). What is needed is *horizontal* thinking – where a section of music is developed by adding and developing sections.[2]

There are a number of ways of going about this, but all of them revolve around thinking *horizontally* - working with sections that contrast and develop, rather than simply stacking sounds onto a single section. The way the sections of a song are organised is called *structure* or *form*. Looking at it this way, there are really two ways to approach this:

1. Know your structure before you fill it. A good example of this is if you know you want to do something roughly mainstream: verse1-chorus-verse2-chorus-bridge-chorus-chorus. You know when you start that you'll need three sections (verse, chorus, bridge) and that one section (verse) will need two variations (verse1 and verse2). You can develop these sections independently in your sequencer. Once you've got them developed to a certain level (maybe drums, bass and main supporting parts) you can arrange them into place by moving and copying the sections. Once the sections are in place you can then make the transitions work and apply the finishing touches. There are other structures and forms you can try using – you can use this approach even if you don't want to compose mainstream music! I'll discuss some structures in future blog posts.
2. See where the music takes you. Using this approach you would develop several sections independently and think about how they fit together later in the composition process. You might use them in a standard mainstream structure (with verses and choruses, as above), or you might do something more complex, or more abstract. This is a more dangerous approach because it's easier to come up with an incoherent song – one that sounds like a "bunch of stuff". You'll recognise this because the sound will sound sprawling, and not feel like it's going anywhere or making sense. It's also easy to fall into this approach as a result of being too lazy (or too self-righteous) to commit to a structure early on.

Whichever approach you take, you have to force yourself to break out of the two or four or eight bar loop. You have to start thinking in *sections*, start thinking about organising these sections to form a song.

And once you've got the hang of that, you can then start thinking about suspense, excitement, contrast, expectation, etc…

-Kim.

[1] For the purpose of this discussion, I'll use the term 'song' as an umbrella term for songs, tracks, works, pieces, etc. This is regardless of whether it has a melody, lyrics, or lead vocal.

[2] I've got some more detailed ideas about the causes of this, but they're beyond the scope of this blog post.

2009/05/09 - Song form

One of the most popular structures is song form. Most popular songs follow a basic pattern:

verse1-chorus-verse2-chorus-bridge-chorus-chorus

This structure works well because there's a fairly even balance between the familiar and the unfamiliar. Done well, there also a clear contour and direction to the song.

1. The first verse and chorus are unfamiliar the first time we hear them, though it's suggested that the chorus is more important than the verse because it's usually louder, fuller, and more stable than the verse.
2. Afterwards the second verse increases the comfort level and familiarity because it's based on material (harmony and rhythm) that we've heard before in the first verse.
3. The second chorus is usually exactly the same as the first chorus, reinforcing its relative importance and further increasing comfort (with familiarity).
4. At this point we've heard two rounds of a verse followed by a chorus. Intuitively we're expecting another verse, but instead we're surprised by the bridge. The bridge usually introduces some fundamental differences (such as new harmony, new rhythm, or new instrumentation) which is surprising and refreshing. It's important, however, that the bridge is "cut from the same cloth" as the rest of the song – that is, it's not so different that it sounds like a new song. Otherwise it'd be so surprising that it'd be jarring to listen to!
5. After the bridge, we return to a double chorus, which is a return to comfort and familiarity, reinforced by the additional repetition.

Of course, there are many variations!

- Many songs have an introduction (an "intro").
- Some songs have a short section that leads into the chorus (sometimes called a "pre-chorus").
- Some songs have three verses.
- Some songs have a two-part bridge (perhaps an instrumental solo followed by a sparse vocal refrain before leading into the final chorus).
- Some songs have a chorus at the beginning.
- Some songs only play half the chorus the first time.

This song form doesn't have to be limited to pop music, or even vocal music though! It works just as well for instrumental electronic music, rock music, hip hop, and just about any other kind of music. Try it!

-Kim.

2009/05/11 - Transition form

Another interesting approach to structure is to think about transition. This is where the prominant feature of the track is the transition between two quite different sections. A good example of this is "The Sunshine Underground" by the Chemical Brothers, which starts out quite slow, relaxed and sparse. By the end, though, it is fast and energetic – with the rhythmic elements sounding at double-time (twice as fast). Another example like this is "Hyperballad" by Bjork.

While both these examples start slow and finish fast, you can explore any kind of transition. For example "Dreaming Your Dreams" by Hybrid starts sounding dark and menacing, but finishes with an uplifting feel – almost joyous and loving!

Feel free to be adventurous in your transitions. Maybe you could start out quick and light but finish heavy and full. Or start out harsh and noisy but finish smooth and clean. Maybe start out electronic but finish acoustic. The possibilities are endless – you are only limited by your imagination!

-Kim.

2009/05/12 - Sonata form

Don't be put off by the fancy name! Sonata form is not just for classical music! It's actually useful for modern music too – especially if you want to do something a bit more adventurous than typical song form. Basically, sonata form consists of three sections:

1. The start (called the *exposition*)
2. The middle (called the *development*)
3. The end (called the *recapitulation)*

The first section (the exposition) presents the main musical themes, the middle section (the development) explores variations based on those themes and the final section (the recapitulation) returns the to main themes.

Classically, a musical theme was most often a melody, but you could use use a groove or a sound as a theme. It doesn't really matter how you do it – so long as the theme has a distinct musical identity. That is, it has to be recognisable.

In the exposition, there are usually two themes. The exposition is divided into two sections – one for the first theme and the other for the second theme. Traditionally these two themes are in different keys, but if your music is more sonic (about sound) than harmonic (about harmony) then you can differentiate the two themes by using different sounds or different textures.

In the development, you can take elements of the main themes and explore variations of them. You can mash them up, combine them, change the style… anything really! You can explore several variations, one following the other. This section is usually best if it's roughly a third to a half of the total length of the song/track.

In the recapitulation, you return to the main two themes. Usually this is almsot exactly the same as the exposition, except that the second theme is made more like the first theme. Classically, this is done by changing the second theme so that it's in the same key as the first theme. In more modern music, you might make the second theme more like the first by using the same sounds or the same rhythms as the first theme.

As with all composition techniques, the best way to learn is to try it out for yourself! Compose several tracks using sonata form and see where it takes you!

-Kim.

2009/05/13 - Making sounds bigger by using compression to manipulate micro-dynamics

Compression is (sometimes) a complex and subtle process. Compression can be used to control macro-dynamics and micro-dynamics. What does this mean?

Roughly speaking, macro-dynamics refers to the volume differences between notes (for example, some notes might be louder than others). Micro-dynamics refers to the volume *envelope* - the differences in volume between different parts of the same note.
The traditional use of compression is to control macro-dynamics – to keep the performance

consistent and even, and to help keep a track in its place in the mix. Think of a vocal performance where some syllables are louder than others, some sustained notes drop off early. Compression can be used to bring a more consistent level to the syllables and words so that they sound more even. It also makes it easier for the vocal to maintain its relationships with the other instruments in the mix – especially a busy or dense mix.

Compression can also be used to control micro-dynamics – the envelope of each individual note. This is most commonly done when using "character" compression on drums – the shape of each drum hit is being deliberately altered. This type of control can be used to make sounds bigger – by flattening a sound's volume envelope, it has more power. Consider the envelope of a drum hit or piano – where the sound starts loud but rapidly decays. Compare this to the envelope of a trumpet or distorted guitar – where the sound stays at the same level for the length of the note. Even though both sounds might start at the same level and have the same duration, the sustained sound is louder and more powerful. Compression can be used to make the drum hit or the piano sound more like that – by either reducing the level of the attack or by increasing the level of the decay.

Practically speaking, this is usually done by using relatively fast attack and release times, coupled with a medium to high ratio and relatively deep threshold.

Just be careful – modern digital compressors can be very powerful, and even free plugins can be capable of completely flattening the sound. This can make a sound very loud and exciting on its own, but cause it to disappear in the context of the mix. This is because the articulation of the sound is greatly reduced (or removed!), so it becomes closer to pure tone. Even though it has more power, its psychoacoustic properties are closer to those of background sounds! This is why busy mixes tend to work better with dynamic, "spiky" sounds, but sparse mixes can be made to sound very full by using heavy compression.

-Kim.

2009/05/14 - Transitions between sections

All this talk of structure revolves around sections – necessarily so, because we're talking about organising a large block of time, and the most common way of doing this is by subdividing into smaller sections.

No matter how you organise your sections, you will still have a skeleton of a song that consists of several sections of various lengths, one after the other. Without any transitions between sections, each section will simply stop as the next begins. The effect will be similar to that of changing channels on a television – abrupt and unsophisticated.

To make a transition between sections work, you must *make something of it*. Articulate it in the music, make a point of the change. Necessarily, there are one of two approaches you can take with each transition – a smooth transition or a contrasting transition.

Smooth transition

A smooth transition is one where the first section smoothly moves the listener into the second section. A common example of this is where the second section is fuller and more exciting than the first section, so the end of the first section has a build up into the start of the second section. Similarly, if the second section is slower or sparser than the first section, the end of the first section might pull back or slow down before entering the second section. The second section might even continue to get sparser in the first few bars.

At an extreme, the transition between two sections might be long enough to be treated as its own section. That is, a whole section in the song is dedicated to transitioning from the previous section to the next.

Contrasting transition

By contrast, a contrasting transition is one where the change from the first section to the second is marked and noticable. It doesn't have to be *sudden*, but it does rely on the two sections being quite different. An example of this might be the sudden jump from a sparse and soft introduction to a song into the full and busy main part of the song. Another example could be a jump from the second chorus of a song into a contrasting bridge section.

At an extreme, a deceptive transition can be used to further emphasise the contrast. An example of this could be where the first section ends by building up as if the second section is louder and fuller, but instead the second section is suddenly quiet and soft. Another example could be where the first section ends by slowing down and pulling back (perhaps even pausing) before the second section suddenly bursts in.

-Kim.

2009/05/15 - Planning mastering

Yes! Mastering should be planned!

I've written in depth about the process of mastering itself, but there's also some important things to consider before you start – especially if you're preparing several songs for publication (public release) together.

Packaging

How many songs will be in the release? Is it a single? An EP? An album? What is the order of the songs? Of you're mastering for CD, will each song have a crossfade transition into the next? Will there be a gap? How long should the gap be?

Distribution

What will the target format be? CD? Internet download? MP3/AAC/WMV? Streaming? Vinyl? Radio? Each of these distribution formats requires a slightly different mastering approach.

Target playback

Who will be listening to the music, and on what system? iPods and computer speakers? Audiophile HiFis? Clubs?

Genre norms

On top of all this, different genres require different approaches – modern rock can be thick and loud, but an acoustic singer/songwriter album needs to be quieter and cleaner. Orchestral music should emphasise dynamics and natural acoustics, but electronic music might need to be clear and solid.

Without considering these factors, you risk compromising your mastering – either getting an ordinary job done or even an inappropriate job done. Mastering requires subtlety and diligence, and it's important that this final processing stage is approached the right way so that your music is presented at its best.

-Kim.

2009/05/16 - What makes structure work?

After experimenting with different approaches to structure, you will begin to vary standard structures and start to think about developing your own approaches to structure. You might start to wonder – what makes structure work? What separates an effective and satisfying structure from an ineffective one?

Contour and proportion

Contour and proportion are about the overall shape of the structure. A structure with good contour is one where the overall rise and fall of tension and excitement makes sense. This means it can be understood by the listener as having a shape that can be followed. The obvious and most common shape is one where the song begins with low excitement, gradually increases to maximum excitement about 2/3 through, and then ends at minimum excitement again. This isn't the only shape that works though! Another shape that makes sense is one where the excitement is greatest at the start and the end, but the middle section is quiet and subdued. Several Juno Reactor songs from their album Labyrinth have a contour like this.

Proportion goes hand-in-hand with contour. At a slightly smaller scale, proportion is about the lengths of each section. A structure with good contour is one where each section is just the right length – not too long or too short. Poor proportion is usually caused by sections that are too long without enough change to keep them interesting. There's no easy rule to help you determine the right length – you have to use your experience and judgement. Shorter sections can be useful for increasing excitement and expectation because they make it feel like the song is moving along at a quicker pace. Longer sections are useful for building tension because the listener is expecting a change that is postponed, or for maintaining and emphasising a hightenened level of excitement during a climax.

Expectation and fulfillment

Expectation and fulfillment go hand-in-hand when approaching structure. Expectation, as you might guess, is what happens when a listener thinks s/he knows what is going to happen next in the music. This expectation is shaped by many factors you can't control, such as personal taste in music and genre norms. A factor you CAN control though, is repeated sequences. For example, if you have three sections – A, B, C – and arrange them in your song as A-B-C-A-B-C-A, the listener will excpect section B to follow. As a more real-world example, standard song form begins with verse1-chorus-verse2-chorus, after which the listener naturally expects a third verse to follow.

Fulfillment is what happens when the listener's expectations are met. In light of the above explanation of expectation, the listener experiences fulfillment when the section that *logically* follows is the section that *actually* follows. Note – this is not *always* a good thing. If the listener's expectations are fulfilled too much the song is percieved to be predictable and boring.
Instead, building expectation but not fulfilling it helps add surprise and interest. It can also support a sense of development and movement in the music. For example, standard song form begins with verse1-chorus-verse2-chorus, at which point the listener naturally expects a verse3. Instead, there is a bridge – new material that surprises and adds interest, and also gives the song a sense of development by increasing the musical scope (adding musical material to the song).

Coherence

Coherence is about the amount of musical material in a song. An easy way of thinking about this is to consider the number of different sections (or melodies, or themes, etc) in a song, as well as the overall length of the song. A song with a high level of coherence will not have much musical material – it might have fewer different types of sections, or its sections might be very similar. Conversely, a song with a lower level of coherence will have a lot of musical material – either more different sections or more variations. Some level of coherence is necessary in music – in order to give the song a distinct musical identity and so that each part sounds like it belongs to the same whole. Too much coherence, however, will make a song boring and repetitive.

Time is also a significant factor contributing to coherence too. Given a certain amount of musical material (say, for example, three different sections), you can increase coherence by increasing the overall length of the song. Similarly, you can decrease coherence by shortening the length of the song. This is an often-overlooked approach. If you're working on a song and you feel like it's too boring and repetitive, try shortening it instead of simply adding new material. Likewise, if you have a lot of musical material (many different sections or musical ideas to organise) and the song is feeling like it doesn't have a distinct musical identity, try making the song longer. This will let the music breathe a bit more – allow the musical ideas to expand and develop.

As with all composition techniques, using them in extreme is usually not the best approach – some judgement is required. And as with all composition techniques, practice is necessary for mastery! You won't get the hang of this first time around – give yourself a few songs to experiment, to develop your own sense for how it all works.

-Kim.

2009/05/18 - Reversed cymbals and the like...

Every time I see someone asking about reversed cymbals, my first reaction is that of disgust… and then I realise that I use them too.

I occasionally find them useful for some songs where dramatic changes between sections are appropriate. The key word here is *dramatic*. Reverse cymbals (and reversed drums in general) as section transitions are dramatic, and aren't very useful for more subdued or relaxed songs.

My usual approach is to use the drums that I'm already using in the song. I'll arrange a few combinations of kick, snare, ride and crash. I'll then render them to an audio file, bring it into the project, and reverse it in place. What that gives me is a few choices that I can use to emphasise transitions or other significant moments in the song. Having several choices means I can vary the sounds, which allows me to get away with using them more often befiore it sounds repetitive. It also means I can use a reversed sound with a strength and character best suited to the "weight" and significance of the transition.

-Kim.

2009/05/19 - On Vibe

What is "vibe"? Abstracting for a moment, it's something about listener's response to your music. More practically, it's the agregate effect of your choices of timing, sound design, composition, mixing, performance, etc. Really, vibe is the result of your taste in music. It's the multitude of tiny decisions you make – often without even realising there is a decision tobe made. You just *do it*.

So how do you create vibe? How do you create music that has a distinctive identity, that *feels* individual, that sounds like *you*?

This comes in two equal parts: you need to learn your tools, and you need to learn *yourself*.

Learning your tools is a topic that's been covered in great depth everywhere. The more of the music production process you do yourself, the harder this is. Develop your composition skills. Practice your instruments. Know your studio equipment inside-out. Become an expert at mixing.

Don't suck at mastering.

Learning *yourself* is more difficult, and isn't discussed very widely. This is about your taste in music, and mostly comes from listening to as music music as possible and making as much music as possible. As a music creator, music is your literature. Listen to as much as you can. Learn to apprecate different styles. Fill your music library with classical, heavy metal, techno, rap, pop, experimental, jazz, etc. The more different varieties of music you listen to, the more you learn about what you like and what you don't. You'll also pick up interesting ideas to try out for yourself. This also helps prevent getting stuck in a rut, or running out of ideas. When all you listen to is one variety of music, you'll find your own imagination becomes limited.

Making as much music as possible is also terribly important. The more music you make, the more opportunities you have to try out different ideas, different approaches, even different styles of music. The more experimentation you do, the better you'll understand what works for you. Try as much as possible to *finish* projects. Don't leave them hanging, and (more importantly) don't get stuck endlessly refining and touching up. You'll learn more by spending a month finishing two or three songs than deliberating over a single song.

How does this relate to vibe? The vibe comes from you. It comes from within. It comes from what intrigues you, what excites you, what moves you.

-Kim.

2009/05/20 - That pumping effect

So you've probably heard enough of that pumping effect by now. Yes, that pumping effect where the whole mix ducks to the kick. Or at least ducks in time with the music. Or at at least some instruments in the mix. Or something.

If you still think it hasn't gone out of fashion yet (or maybe you're waiting for it, like hard pitch quantisation, to come back in fashion) you might be wanting to brush up on your technique. Or you might want to explore some different techniques to come up with something new and different. Or maybe you're just a sucker for punishment. There are four main ways to approach this:

Straight full-band compression

This is how it was done in the old days – a big kick drum in the mix, and a compressor on the mix bus. For the most distinct effect, the kick should be just loud enough that it should be the highest peaking sound in the mix. You can check this by looking at the peak meter on the mix bus. The kick is loud enough when the peak meter jumps up whenever the kick plays. The mix bus compressor should then be set so the threshold is just low enough that it is triggered by the kick and nothing else. With a high ratio, fast attack and medium release, you should be able to hear the effect pretty clearly. For a stronger effect, use a higher ratio.

Full-band compression with pre- and post-emphasis

For a more dramatic effect than straight full-band compression, insert an EQ on each side of the mix bus compressor. That is, one EQ before the compressor and one after. Set the first EQ to boost the bass significantly (for example, a low shelf +9dB at 150Hz) and set the second EQ to make the opposite cut (for example, a low shelf at -9dB at 150Hz). Coupled together, you should hear an unchanged frequency response. However, what the compressor "hears" is a mix with a LOT more bass than usual. With a strong kick, this can make the compressor behave in an exagurated, slightly unnatural manner. Which might just be what the doctor ordered.

Sidechaining

The two above techniques will fall down, however, on mixes with very strong basslines and relatively weaker kicks. This is expecially so where the bassline is voiced very low and is fairly constant (always sounding, without rests). In these cases, you might want to try true sidechaining. This is where the compressor on the mix bus is operating on the whole mix, but is only "listening" to a separate feed which has the kick drum on its own. Not all compressors can function in this mode, and the ones that do can be fiddly to set up and control.

LFO modulation

Another approach is to use an effect with a tempo-synced LFO. You'd have to set the LFO period to a crotchet (a quarter-note), and its shape to a rising saw. Then assign that LFO to control the gain of the plugin. Volcano from Fabfilter[1] can be configured like this. Apparently Camelphat from Camel Audio is also capable of this, though I haven't personally tried it myself. The advantage of taking this approach is that being able to adjust the shape and depth of the LFO gives you different, and sometimes more intuitive, control over the sound of the pumping.

While it's often written that the effect is always produced by side-chained compression, there are other ways of achieving it. Sometimes the obvious solution is not the most appropriate one. And sometimes trying something new can take you places you never thought you'd go…

-Kim.

[1] Disclaimer: I have a professional relationship with Fabfilter.

2009/05/21 - Variations on that pumping effect

Sidechained compression is another technique that has several uses. It can be used on individual tracks to create a dramatic ducking effect without affecting the rest of the mix. You could trigger compression on a synth pad with the kick or snare for dramatic effect. It can be used as a special effect vocals. This is possible both ways – to duck the vocal when the kick is played, or to duck other instruments while the vocal is heard. It's now common in some circles to use kick-triggered sidechain compression on a synth bass, so that the dramatic pumping effect almost sounds like the attack time is being automated. This focusses the effect so that the rest of the mix remains intact. It's also common to trigger a side-chained vocal compressor with a high-passed version of the vocal, to reduce the sibilance (harsh "ssss" sounds).

The real fun starts when you take these techniques beyond the ordinary pumping effect. Pre- and post-emphasis is a handy technique that is applicable beyond compression. You can use it any time you want a plugin to respond to a skewed tonal balance but without changing the balance as it's heard. One approach is to use this with saturation. By temporarily boosing the low frequencies or high frequencies, you can focus the saturation on that range. By scooping the mids you can sometimes make the saturation smoother.

Sidechained compression is another technique that has several uses. It can be used on individual tracks to create a dramatic ducking effect without affecting the rest of the mix. You could trigger compression on a synth pad with the kick or snare for dramatic effect. It can be used as a special effect vocals. This is possible both ways – to duck the vocal when the kick is played, or to duck other instruments while the vocal is heard. It's now common in some circles to use kick-triggered sidechain compression on a synth bass, so that the dramatic pumping effect almost sounds like the attack time is being automated. This focusses the effect so that the rest of the mix remains intact. It's also common to trigger a side-chained vocal compressor with a high-passed version of the vocal, to reduce the sibilance (harsh "ssss" sounds).

LFO modulation is where things get really interesting. Using a tool like Fabfilter Volcano[1], you don't even have to have the kick playing in order to hear the pumping effect. You could vary the modulation strength and shape – even use different pumping shapes in different parts of the song to emphasise the different levels of excitement or tension. You could also experiment with coupling the volume modulation with modulation of other effects – filtering, overdrive, chorus/flanger, stereo widening, etc.

Remember – interesting things happen when you experiment and explore different techniques! Keep an open mind and you'll be sure to develop your skills and knowledge.

-Kim.

[1] Disclaimer: I have a professional relationship with Fabfilter.

2009/05/25 - About monitoring environments

Monitoring environment is critical to production and mixing. You must be able to accurately hear and interpret the sounds you're working on in order to make reasonable decisions. Achieving a good monitoring environment, however, is not as simple as buying the most expensive speakers you can – there are several components to a good monitoring environment: The space, the speakers, and headphones. Ideally, they should all work together and complement each other.

The Space

The space you listen in is just as important as the speakers. There are many different types of acoustic spaces, and if you want to get the best out of your space you should try to understand the relevant acoustic properties. Spaces generally have three broad properties that you should pay attention to:

- The size and dimensions of the space
- The surface coverings
- The placement of objects within the space

The Speakers

The speakers are the most obvious component of a monitoring environment, but not necessarily the most important. It's also not simple to choose speakers. For example, simply getting the largest woofers you can afford is not always the best approach – speakers with larger woofers tend to be focus their energy and accuracy on the lower frequencies, sometimes at the expense of accuracy in the upper mids. Conversely, smaller speakers might be more accurate in the upper mids but weak in the bass, making it more difficult to correctly judge mix decisions for the kick and bass – critical in modern dance music.

Headphones

Headphones are also a part of a well-balanced monitoring environment. They offer a different listening perspective to the speakers in your room. Generally, headphones allow more detailed and focussed listening, which makes them ideal for spotting problems in recorded audio (such as background noise or interference). The drawback is that the sound is generally drier and wider than when listening with speakers, making them inappropriate for judging front-to-back depth. If you choose well, however, you can use your headphones to compensate for weaknesses in your speakers – either using bright headphones alongside muddy speakers or deep headphones alongside thin speakers.

I intend to address each of these topics in more detail in future blog posts.

-Kim.

2009/05/26 - Different types of reverbs

Just quick run-down of some common types of reverb:

Hall

This is the most common type of reverb. As the name suggests, hall reverbs are usually designed to simulate the kind of reverberation effect produced by large halls. A hall reverb is usually a good choice for adding a three-dimensional ambience to your mix. Good hall reverbs tend to fill out the back of the mix, adding depth without crowding the forground.

Room

Room reverbs are similar to hall reverbs in that they are usually designed to simulate the natural sound of an acoustic space. Unlike halls, rooms are (obviously) smaller spaces. A room reverb is good for adding realism to instruments that have been recorded with very close mic positioning or direct injection. Guitars and drums are likely candidates for room reverbs. A good room reverb will give you the sense that the instrument is being played in a real acoustic space.

Plate

Plate reverbs simulate an earlier method for generating reverb – by injecting sound into a large hanging metal plate and letting it reverberate. Plate reverbs have a similar shape to hall reverbs, except the sound is usually denser and flatter (two dimensinal). Plate reverb is great for adding length and size to a sound without making it sound distant or small. Snare drums and lead vocals tend to work best with plate reverb.

Spring

Spring reverbs simulate a method of generating reverb that is commonly built into guitar amplifiers – by injecting sound into metal springs and letting them reverberate. Spring reverbs tend to sound bouncy and lo-fi. Personally I can't stand them, but you might find them useful on guitars. Or vocals if you're feeling particularly sadistic.

Inverse

Inverse reverbs simulate a "backwards" sound by generating a reverb that increases over time, isntead of decreasing like other reverbs. These are not based on any sound in reality and are sometimes useful for special effects or unnatural ambiences.

-Kim.

2009/05/27 - The space

The space you listen in is just as important as the speakers. There are many different types of acoustic spaces, and if you want to get the best out of your space you should try to understand the relevant acoustic properties. Spaces generally have three broad properties that you should pay attention to:

- The size and shape of the space
- The surface coverings
- The placement of objects within the space

Size and shape

The size and shape of the studio are a critical factor in determining the general acoustic character of the room. Parallel walls create standing waves, which skew the native frequency response of the room. Unfortunately, most rooms have several parallel walls! The more square a room is, the greater the problems with standing waves. Larger rooms have standing waves at lower frequencies, which interfers less with the audible audio range. Conversely, smaller rooms have standing waves at higher frequencies, which interfers more with the audible audio range. The ideal size is a trade-off, however, because larger rooms bring other problems, such as reverberation. Also, the ideal room shape wouldn't have any parallel walls, but you might have to settle for less if you're not in a position to design a new room (or inherit a previously-designed acoustic room)

Surface coverings

The type and arrangement of surface coverings will affect the reverberation characteristics of the room. Uneven surfaces (such as bookshelves) will "break up" the reflections and make the reverberation smoother (which is less distracting). Soft surfaces (such as foam or fabric) will absorb sound and reduce the reverberation time and level. Again, correct treatment of surfaces is a trade-off. Too much absorption will makes the room sound dead and unnartural, which may encourage you to create mixes that are more dense and washed out.

Placement of objects

The placement of objects within the space also affects the sound of the room. Large objects can help absorb lower frequencies that surface coverings can't absorb. This can be used to make the room less boomy. Objects can also diffuse the reflections in the room, helping to make the reverberation smoother.

-Kim.

2009/05/28 - Speakers

The speakers are the most obvious component of a monitoring environment, but not necessarily the most important. It's also not simple to choose speakers.

Full-range monitors

Full-range monitors usually have large woofers and are designed to produce sound down to low frequencies (in addition to high frequencies). These monitors try to reproduce the broadest range of sounds. Because of this, these can be attractive as "first" monitors. Be aware though, that there's no free lunch. Reproducing those lower frequencies requires large cones, which are good at reproducing lower frequencies but aren't so accurate at higher frequencies. The top of the frequency range is usually covered by the tweater, which is good at reproducing high frequencies but not as strong on lower frequencies. This approach results in a speaker that is strongest at high and low frequencies but might be weaker in the mid-range.

Smaller speakers

Another approach to the size/frequency trade-off is to use a medium-sized woofer with a tweater. This approach focusses the strength and accuracy of the speaker in the middle and high frequencies. This is good for making a lot of mix decisions, because the mids and highs is where the most instruments are playing together and where its most important to get the balance right in the mix. The drawback, of course, is that these speakers are weakest in the bass. This can be a particular problem when working on modern electronic music, where the kick and bassline are extremely important.

Subwoofer

A common solution to the problem of smaller speakers' bass response is to add a subwoofer. This is a third speaker focussed on the lowest frequencies. This has the potential for a more accurate approach across the whole frequency range. The drawback, however, is that it's easy to misconfigure the subwoofer (usually by making it too loud). The way the subwoofer works with the other speakers depends greatly on the room their placed in, meaning the configuration is very much up to you (or whoever configures your room for you). It's common to hear large amounts of bass as pleasing or exciting, making it difficult to resist the urge to configure the system to sound exciting instead of accurate. Another problem more common with cheaper subwoofer-based systems is that the front speakers are too small to reproduce the lower mids (which are poorly compensated for by the subwoofer) , and the subwoofer is too small to accurately reproduce the lowest frequencies anyway.

Of course, a well-rounded monitoring environment consists of more than one set of speakers so that the weaknesses in a single set doesn't become a weakness in the whole monitoring environment.

-Kim.

2009/05/29 - Headphones

Headphones are a part of a well-balanced monitoring environment. They offer a different listening perspective to the speakers in your room. Generally, headphones allow more detailed and focussed listening, which makes them ideal for spotting problems in recorded audio, such as background noise or interference. They're also essential for recording acoustic instruments such as vocals, guitars or drums.

The drawback to using headphones is that the sound is generally drier and wider than when listening with speakers. Personally I find they can be misleading for judging dynamics too – the difference between levels seems to be smaller than when listening to speakers. These factors make it more difficult to use headphones for judging front-to-back depth.

What headphones *are* good at, however, is zooming in on audio. Headphones can be great for hearing details that you might otherwise miss with speakers – such as rumble, hiss, crackles, breaths, background noise, distortion etc. Headphones can be excellent for surgical correction and cleaning up. They can also be useful for judging subtle distortion when using limiters and saturation to reduce headroom in mastering.

If you choose well, you can use your headphones to compensate for weaknesses in your speakers – either using bright headphones alongside muddy speakers or deep headphones alongside thin speakers.

-Kim

2009/06/01 - Dragging out the tools

There's only two reasons to drag out the tools:

(1) To play with them because you don't know what you're doing.

It is necessary to do this in order to learn your tools. Whether your new tool is a reverb processor, a compressor, a guitar, a new set of speakers or even a new studio chair, you need to spend some time with it alone without the pressure and obligation of working for a client. Pull apart some old projects – grab the drum tracks, the vocals or guitars and try to recreate the same sound using the new tools. Twist some knobs, explore the range of the tool. Explore it, see what it can do. This is important and necessary – just don't do it in front of a client!

(2) Because it's the right tool for the job.

This is what you do when you're on a job. You pull out your tool because you already know it's the right tool for the job. You already know what settings you're going to use and what the result will sound like – before you've plugged anything in. You get here by spending a lot of time doing (1). It's called practice.

-Kim.

2009/06/02 - Mixing in mono

There are several different approaches to panning and stereo width in mixing.

Personally I leave panning until the end of the process. While I'm recording and mixing I have everything panned centre. The only exception to this is tracks where the stereo component is a key characteristic of the sound, such as stereo synth pads or double-tracked guitars. It's only when I've finished adding parts to the mix that I then pan the parts.

What this does is artificially make the mix more dense as I'm working. This forces me to pay careful attention to how each part interacts with all the other parts (and the whole context of the song). It also helps me to use foreground and background more effectively. Being unable to pan means there is only a limited sonic space available for the foreground, so I can't use the left-right space to jam more sounds in (which ends up sounding mushy and crowded anyway).

It's almost always critical that mixes hold up in mono. Sometimes a mix might be summed to mono in playback (low quality streaming websites, poor FM radio reception, some television), or sometimes a listener will only hear *one side* of your mix (imperfect listening position, speakers placed in different rooms, shared earbuds). Even in the right listening space, most people don't pay much attention to panning.

-Kim.

2009/06/03 - The right vocal level

It's not always easy to find the right level for a vocal. It's common to feel the need to mix the vocal low (especially if it's your own vocal!), burying it among the other instruments. It's also common to mix the instrumental backing first, and the find that either the vocal rides unnaturally over the top or is unintelligable. Sometimes it might seem that the vocal doesn't fit in – no matter what EQ, compression or reverb you use.

The trick is to acknowledge that the vocal is the most important part of the song, and the most important part of the mix. That means it should be in the foreground. If you *start* with the vocals, you are free to use as little EQ as necessary – ensuring you have as natural a sound as possible.

Then build up your mix around the vocal, bringing in instruments in order of importance. In a mainstream pop/rock mix this order would be something like: drums, bass, guitars/synths, backgrounds. The relative levels of each group of instruments should be determined by the style of music. For a lot of pop music, the vocal should roughly be at the same level as the snare (with the sibilance at roughly the same level as the hihats). Bass is next and guitars are usually further behind, but in rock the bass is further back and the guitars are closer to the front.

Reverb should be added later – it's not a fixer! The vocal should have the right tone, dynamics and correct level relative to the rest of the mix *before* any reverb is added. The reverb merely adds ambience and makes the vocal sound less naked. If the vocal doesn't fit in the mix without reverb, then adding reverb won't help.

-Kim.

2009/06/04 - Producers and "producers"

I'm an old curmugeon.

Something interesting has been happening to music production. More specifically, something interesting has been happening to the role of the music *producer*.

I commonly see the term "producer" to refer to various roles, usually something like "the guy at the computer" or "the guy who does the mixing", or even "the guy who makes the beats". Maybe a "producer" is simply someone who produces music?

Traditionally, the producer is someone who works with a band and is responsible for the creative vision of a project (usually an album). The producer might perform any of the following tasks:

- Help choose band members or additional studio musicians
- Help with instrumentation and arrangement of songs
- Maintain the mood, esteem and motivation of the band members
- Communicate the creative vision for the project to third parties such as studio musicians, mix and mastering engineers, marketing and promotion teams, record label staff, etc
- And others…

Nowadays though, modern technology has allowed recording studios to be owned and run by almost anyone. Now people with creative vision for music projects are likely to skip working with a band. Instead they'll set up their own computer studio and write and record their own music. Now all of a sudden there's a single person who is a composer, musician, recording engineer, mix engineer, studio owner, promoter, etc.

Sometimes this "producer" will outsource to cover gaps in their own knowledge and abilities. Weaknesses in songwriting or composition skills can be covered by doing remixes or covers. Weaknesses in singing or playing instruments can be covered by bringing in musicians. Sometimes people try to cover weaknesses in engineering skill are by purchasing more expensive gear. Either way, the lines are blurrier now, and the term "production" has grown to include a number of newer roles.

In todays smaller studios, there is production implied when people speak of recording or mixing. Likewise there is often recording or mixing (or even composition) implied when people speak of production.

-Kim.

2009/06/09 - Stability

Stability is the effect of a number of elements in the song all contributing to a sense of predictability and comfort. Some techniques that can produce a sense of stability are:

- Simple, regular rhythm; or repetitive or predictable rhythm (Not just drums, but rhythm of other elements too).
- Unchanging or predictable sonic texture.
- "Easy" chord progressions.
- Consonant harmony (Consonance is the opposite to dissonance)

Stability comes from repetition, predictability, and simplicity. Two common uses of stability are:

Exposition of material. This is the introduction or presentation of new musical ideas. Stability is useful for exposition because material is easier to remember when it is easy to digest (the mind remembers material better when it doesn't have to work hard to interpret or decypher it).

Conclusion. Stability is usually a desirable attribute for a conclusion (ending) section because it tends to resolve expectations (rather than create them), that is – it makes it easier to "finish" questions, rather than ask new ones. Stability is very effective in resolving tension.

The flip side of stability is **instability**. This is the opposite of stability – where tension is easier to create, where the listener has to work harder to understand the music. Instability is often where things get much more interesting as well – this is where the music develops and veers into new territory. Instability is usually unpredictable and unsettling – there are more surprises, new sounds, and sometimes unpredictable patterns.

As a composer (or producer), it should be farily obvious that it's a good idea to aim for a balance of stability and instability within a song. Beyond mere balance, think about deliberately controlling the level of stability throughout the song so that it enhances the contour of the song. For example, pop songs have most stability during the chorus. This reinforces the chorus as the most important and memorable part of the song. It makes it easier for the listener to understand and memorise. Conversely, the bridge is usually less stable. This adds interest and catches the listener's ear. It also sets up the following chorus – making the stability of that chorus more welcome. Sometimes the introduction (intro) of the song is very unstable – again, this is to catch the ear before diving into the relative stability of the verses and choruses. While instability is useful, too much will make the song difficult to understand and listen to. Conversely, too much stability will make the song boring and uninteresting.

-Kim.

2009/06/10 - EQ on the mix bus

As I've written previously, there are compelling reasons for and against using processing on the mix bus. What I recommend against, however, is using static EQ on the mix bus. The reason for this is that pure EQ (discounting EQ with added saturation) has the same effect regardless of whether it is processing single tracks or groups.

For example, if you apply a +6dB boost at 100Hz on your mix bus, this is no different to applying the same +6dB@100Hz on *every single track* in your mix. This is a blunt instrument! If you are applying EQ like this, it is usually to bring out a particular character in certain instruments (such as the kick and bass in this example). If you're mixing, however, you already have access to the individual tracks and can apply EQ in a much more targeted and specific manner. In this example, a better approach might be to simply raise the level of the kick and bass, or apply a low-end boost to the kick OR the bass. Avoiding mix bus EQ would also avoid boosting any mud or rumble in the other tracks, helping you achieve a cleaner sound.

There are a couple of scenarios when applying EQ to groups of instruments may be appropriate: Group EQ and automated EQ.

Group EQ is useful when you have several tracks that sound very similar (such as several takes of a single instrument) and you wish to apply the same EQ curve to all those tracks. I often do this when I have guitar or vocal stacks – multiple takes of the same instrument layered for a thicker sound. Applying EQ to the group saves time because I don't have to set the EQ on each track, and it saves CPU/DSP because there's only one instance of the EQ (instead of several).

Automated EQ might also be useful on the mix bus as a special effect – you might have dramatic changes in your song which include a build-up with the whole mix high-passed or low-passed. Applying this automated EQ on the mix bus is easier and more dramatic than applying it to individual tracks.

-Kim.

2009/06/11 - Familiarity

"Familiarity" is a way of describing the effect that a section of music has on its listener when the listener recognises something in the music. Of course, the opposite of familiarity is unfamiliarity – when the listener does *not* recognise something in the music.

Degrees of familiarity refer to the idea that we don't just have "Familiar" and "Not Familiar" – there are many levels in between. For example, we could speak of "Not familiar", "Partially familiar", and "Completely Familiar".

When a section of music is not familiar, it means the listener does not recognise it – this is usually because the material is new (has not been presented earlier). This can create uncertainty, but can also have an effect of opening up expectations for a piece – for example, a short unfamiliar section right after the introduction of a piece can give the impression that more will be revealed, or that the unfamiliar material will be developed (or revisited) later in the piece.

When a section of music is partially familiar (a bit familiar), it is often because is bears some resemblance to previous material but it is changed, or developed. Some examples could be -

- A new melody that shares the same rhythm of a previous melody
- A bassline that matches a previous chord progression
- A synth pad that's been vocoded with the chorus vocals.

Slight familiarity can be powerful when used effectively, because it is a bridge between something known and something unknown. It can be used to soften a transition to new material. It can be used to give hints of where we came from, of where we're going to. It can be used to make a new section feel more like it "belongs" to the piece.

When a section of music is completely familiar, it means the listener has heard it previously in almost exactly the same form. This is usually the result of literal repetition (repeating some material without changing it) and can be used to reinforce some particular material (like the chorus of a pop song). Complete familiarity can also be desirable at the end of a piece – at the conclusion. This is very similar to my mention of stability at the end of a piece.

Layers of familiarity is what happens with some parts playing familiar material, and some parts are playing unfamiliar material. Layers of familiarity are interesting for the same reasons as partial familiarity – it creates a crossover between known material and unknown material.

The concept of familiarity is not as concrete as some other aspects of music theory (such as chord progressions, voice leading, or filter sweeps). This is not meant to be the focus for a composer, and it doesn't really make sense to study it on its own.

It's something to keep in mind as you compose, something to think about in between figuring out how many times to repeat that drum loop and when to bring in the next synth line.

Familiarity can be linked with Stability, and Expectation. It's important to understand, however, that while these concepts may often correlate (when familiarity is high, often stability is high as well) they are not the same, and interesting results can be achieved by mixing them in unusual ways.

-Kim.

2009/06/13 - Devil Gurl and MAutoEqualizer

I've just published Devil Gurl – a harder electro-rock from MEGALOVE. This is the fourth in this series, and the first with live drums (performed by our drummer who plays at our gigs).

This is the first project that I've started to include MAutoEqualizer in my mastering workflow. Basically this is an EQ that analyses some reference material along with your own track, and then applies appropriate equalisation to make your track have the same tonal balance as the reference material. Unlike some other similar plugins, MAutoEqualizer uses Fast Furier Transform for analysis only – the actual signal processing is performed using parametric equalisation (minimum phase and linear phase algorithms are both available).

I was initially sceptical about this approach – surely tonal balance is a matter of judgement and creative direction? Well, in mixing it is. Mixing is a creative endeavour. My approach to EQ in mastering though, is to adjust the tonal balance of the audio so that it favourably compares with my commercial references. While this requires considerable skill, it does not require a great deal of creativity.

The idea of having software algorithms do this task for me is actually quite enticing. Regardless of whatever mastering skill I've developed, I still find it to be a chore – boring and tedious – precisely because it is not very creative. Having said that, MAutoEqualizer doesn't perform miracles. Getting a good result requires two things:

1) **A good mix**. Automatic equalisation isn't a magic fix-it. All it does is adjust the overall tonal balance. Mastering EQ is extremely limited in its ability to change the balance between instruments or to change the tonal characteristics of individual instruments.

2) **Generous and appropriate analysis material**. I find the best results require giving MAutoAnalyser several different songs to analyse for reference. This ensures the target tonal balance is more "average" and doesn't contain any idiosyncratic irregularities specific to a single song. As has been my practice for years, I choose 3-4 reference songs in a similar style, but from different artists and albums. I just line them up on a track one after the other and let MAutoEqualizer listen to them in one pass. Similarly, it's important for MAutoEqualizer to analyse the whole of the song being mastered.

-Kim.

2009/06/15 - Variation and development

Put simply, **variation** is when you take a bit of music (it could be one bar, one instrument, or even a whole section) and you *change it*. More accurately, you would probably make a copy of that bit of music, and change the copy. The key is that the changed copy still bears some resemblance to the original (ie, we're not talking about transforming it into something unrecognisable).

Easy.

What then, is **development**? Development is a more sophisticated version of variation. If we define variation as "making some change", we could define development as "using a defined process to make a variation". At first glance, the difference may look like one of semantics, but I assure you it's more than that. Before we discuss why we'd want to use development, I'll give you some examples so you know what I'm talking about.

Example one:
Let's say we have a drum pattern. Let's say we want to vary the position snare drum. If we were to make a variation, we might "randomly" move the snare drum hits around, perhaps inserting some or removing some. The variation will not actually be "random" – we'd be changing the snare hits according to what we think sounds good. I use the term "random" because it helps illustrate the difference between variation and development.

If we were to make a development of the drum pattern, we would use a defined process to alter the snare drum hits. Defined process? Well, we could do something like move all the hits one sixteenth of a bar earlier. Or we might gradually increase the density of snare drum hits (one hit in the first beat, two hits in the second beat, … , four hits in the fourth beat). Or we might make the velocity (volume) of each snare drum hit increase as they progress throughout the bar. Or we might do all three.

Example two:
Let's say we have some melody, and make a copy and we want to change the notes on the copy. If we were to make a variation, we'd change the notes "randomly" – according to whatever we think sounds good.

If we were to make a development, we might do something like transpose each note one step higher than we transposed the previous note. Or we might change all upwards jumps to equal downwards jumps (and vice-versa). (You might want to measure "steps" in your favourite scale, to avoid getting "wrong" notes. Or you might like the sound of the "wrong" notes!)

The difference between variation and development, is that for development we're using a defined process over a period of time. You may also choose to think of it as a *repeatable process*. We could take the process that we used, and apply it to some other tracks, or another section.

From those examples, you might already be thinking about some ways in which development may be useful as an alternative to variation.

One advantage (that I've already mentioned) of development over variation is that you can use some process, and then apply the same (or a similar) process to other bits of music. For example, you could perform some development on a drum track during a bridge section, and then do the same thing on the bassline, or the chords, or whatever. Or you might make a development of the main melody, then perform and inverse or opposite development on the bassline.

Of course, multiple develpment doesn't have to be just in parallel – you could do them one after the other. For example, you might have a *really* dense drum pattern. You might have it plain once, then for the next repeat you could use some process to remove some hits. Then for the next repeat perform the same (or similar) process *on the previous development*, and keep doing that until the drum pattern is empty.

Another interesting approach could be to apply a similar process across *different lengths of time*. For example, you could come up with a process to let you thin out a drum pattern rapidly – so that at the start of the bar it is complete, but by the end of the bar there is nothing left. You could then apply a similar process to the bassline, but *across two bars*. Then do the same thing to the pad, but *across four bars*. Then the meldoy, *across eight bars*…. or something like that.

While I'm using typical tradtitional western music constructs (notes, metric rhythm, drums, melody, chorus, etc) for examples, that these principals are appropriate to almost *all* kinds of music.

-Kim.

2009/06/16 - Multiband compression

Multiband compression is a complex and subtle tool. Compression itself is one of the most complex single processes commonly applied in mixing. Multiband compression multiplies that complexity because it applies several compressors in parallel, each processing a different frequency range of the audio. Because of the way the audio is split by frequency, multiband compression is best suited to complex audio with varying dynamic behaviour across the frequency range. Typically, this would be a full mixdown (either on the mix bus, or in mastering).

Multiband compression is best used for one of two purposes – surgical problem solving, or subtle leveling.

Multiband compression is ideally suited to some kinds of problem solving because it allows compression to be applied to a specific frequency range without altering the rest of the audio. For example, a mix with uneven bass guitar playing could be improved by using multiband compression to reduce the dynamic range of the low frequencies. Another example could be a mix where the vocal has not been compressed appropriately and alternates between being too quiet and too loud. Depending on the mix, multiband compression can be used to even out the vocal in relation to the rest of the mix.

In these examples, multiband compression would be used at the mastering stage only if it's impractical to revisit the mix. Of course it is better to fix these problems in the mix (or even earlier) if at all possible.

The other common use for multiband compression is for subtle leveling. Rather than using a single band to solve a specific problem, all bands are activated and are gently riding the gain. This approach works best on weak mixes that are not balanced very well. It improves the overall tonal balance and dynamic behaviour of the mixdown in a more subtle and less damaging way than full-band compression. Again, this approach is appropriate if it is impractical to return to the mix.

As always, the earlier these problems can be addressed, the more power you will have to apply the appropriate solution. Don't wait until mastering to fix things that should be addressed in the mix! Likewise, don't wait until the mix to fix things that should be fixed in recording or even composition!

-Kim.

2009/06/17 - Expectation

Expectation is what happens when a listener anticipates a future event in the music. Put another way, it is when the listener has some idea of what will happen in the future of the music.

This is usually achieved through repetition. For example, if I present a pattern:

A A B C, A A B C, A A B ?

The listener would probably expect C to take place next (where the ? is).

Expectation can be a powerful tool for manipulating the experience of the listener. By being aware of patterns and the degree of repetition in your work, you can have a greater understanding of the expectations that the listener has as they are listening to the piece.

If you have a section of music where you think the listener will be expecting something in particular, there are two things you (the composer) can do:

- Fulfill their expectation.
- Deny their expectation.

Of course, this is a very simplistic view, but will serve for the purposes of discussion. Feel free to explore the grey area in between on your own!

There's not much to say about fulfilling the listeners expectation, only that it usually increases stability and reinforces familiarity.

Denying the listeners expectation is much more interesting, because this is where the music breaks the pattern and surprises the listener.

There are two special cases of expectation denial which are particularly interesting: that of *pushing back* the expected material, and that of *pushing forward* the expected material.

What do I mean by this? Let's start with *pushing back*. This is when the expected material occurs later than the listener thinks it will occur. Using the above example, *pushing back* might look like this:

A A B C, A A B C, A A B B C

Notice that on the third repeat, the position where the listener would expect a C is actually occupied by a second B, and the expected C actually occurs one unit time later.

Pushing back usually has the effect of suspending the progress of of piece. If done well, *pushing back* can **increase** the expectation of the listener – that is, make the expectation stronger. This is common in pop music where there is a bar or two of something between the end of the second (or third, etc) verse and the chorus. In this case, it makes the listener want the chorus more, and feel more satisfied when they actually get it.

On the other hand, *pushing forward* means giving the listener what they expect, but earlier than they expect it. Again, using the above example, *pushing forward* might look a little like this:

A A B C, A A B C, A A C

Notice that during the third repetition, the listener expects the C, but actually gets it one time unit earlier.

Pushing forward usually has an effect of speeding up the pace of the music, and can also increase the excitement of fulfilling the expectation.

-Kim.

2009/06/18 - Monitoring gain staging

The reason the commercial references are so loud is that they have very little headroom – the average level is so high that there's not much room for the peaks (which have been squashed down). When mixing, however, you shouldn't worry about headroom on the mix bus. You need to give yourself enough headroom that you can focus on the task of mixing – getting the balance between the different sounds right. Wait until mastering before you tackle the "mastering loudness" problem.

If you try to mix at commercial levels (with extremely low headroom) by using a limiter on the mix bus, you'll have an extremely difficult time of it because every small change you make to a sound will change the other sounds in noticeable and largely unpredictable ways. For example, if you turn up the vocals, the bass might become quieter (because *everything* becomes quieter). Noticing the bass, you turn it up, which might make the whole mix more distorted (particularly if you're using saturating processors on your mix bus). It also causes problems because setting a channel to solo causes it to sound very different to how it sounds in the mix (because it's being modulated by the other channels).

To make sure you have enough headroom at mixing, simply *turn your speakers up*. Turn the volume up much higher than you would have it for regular listening. Don't use any processors on your mix bus. Your mix will end up "quiet" (because it's much further from 0dBfs than your commercial references), but don't worry. Mastering is a separate process, and one of the purposes of mastering is to bring the overall volume up to its final level.

By doing this, you are free to focus on the mix.

-Kim.

2009/06/20 - Peak vs RMS

Peak levels are the highest digital values that are in the waveform as it exists in the computer (or other digital equipment). While peak levels aren't directly related to how we hear the sound, they are crucial for correct gain staging in digital gear. Most critically, peak levels must not reach 0dBfs when recording or when preparing audio for distribution.

There are many factors other than raw level that influence the way we hear sound, which is why peak level alone isn't directly related to how we percieve sound. For example, we perceive longer sustained sounds to be louder than short transient sounds. Likewise, we percieve sounds in the upper mid frequencies (around 2.5kHz) to be louder than sounds with extremely low or high frequencies (such as 100Hz or 10kHz). So a 100ms sound at 100Hz will sound much quieter than a 1000ms (1s) sound at 2.5kHz, even if they're both at the same peak level.

RMS is a way of measuring audio levels that more closely reflects how we perceive sounds. Roughly speaking, RMS measurements average the level over time, so that short transient peaks don't have as much influence as longer sustained sounds. This way the RMS measurement is less "jumpy", and better represents how the audio sounds to us.

-Kim.

2009/06/22 - Theory vs creativity

Some people seem to believe that theory and creativity are somewhat at odds with each other. It seems the typical line of thinking is that a person who knows no theory is free to compose whatever she hears in her head. Unhampered by preconceived notions of "right" and "wrong", the composer can get as close as possible to her ideal.

On the other hand, a composer who has studied theory is supposedly hindered by what is supposedly "right" or "wrong". Perhaps she came up with an idea, but (consciously or subconsciously) rejected it because it didn't fit within certain "rules".

This is not necessarily incorrect – each person composes in their own way – but it doesn't have to be like this.

There are two kinds of music theory that a composer may draw upon - general composition theory, and "genre" theory.

General Composition Theory is independent of style, instrumentation, size, etc. It is applicable to all composition. The important thing to remember is that **it is not a set of rules**. General composition theory is a set of tools, a set of techniques. It's not what's right or what's wrong. It describes common practices and their effects upon the listener. It's up to you, the composer, to decide what's "right" and what's "wrong" – depending on your own context. Feel free to ignore them, disobey them, break them. Feel free to embrace them, extend them, use them to make your own set of tools.

In my experience, learning general composition theory hasn't limited my creative strength. On the contrary, it's definately increased it. I still come up with whatever ideas I want, and I still arrange them according to whatever sounds good. But in addition to that, general composition theory has given me ways to develop and extend my ideas to form works of much larger scale. Being conscious of the tools and techniques has allowed me to arrange and develop the material so it is stronger, more complete, and has greater impact.

It's commonly said that creativity is 10% inspiration and 90% perspiration. Theory helps you get the most out of that 90%.

Remember though: as the composer, it's up to you to learn from this. It's up to you to integrate this knowledge into your own style of composition, into your own set of techniques. **You have to own it**. Remember: This is art, not science.

The other kind of theory is **Genre Theory**. This includes "how to write trance", or "how to write the perfect pop song". This also includes traditional theory as you may know it – the rules upon which "classical" music is composed. Genre Theory is much more often presented as a set of rules – and rightly so. If you want to compose awesome Trance music, you'd better get out that TR909 and stacked supersaw lead (or whatever the kids are using these days). If you want to compose classical music, you'd better brush up on your scales and cadences.

Unfortunately, many people incorrectly believe all theory is "classical" theory. I imagine this comes from some lower-school education, where this was the only type of theory that was tought. Unfortunate, really.

-Kim.

2009/06/23 - Compressing vocals

Vocals are usually compressed to make the dynamics even – that means that all the syllables are roughly the same level. That makes it easier keep the vocal balanced with the rest of the instruments. Otherwise the soft syllables will be lost in the mix but the louder syllables will stick out too much.

Start with highest threshold, highest ratio, fastest attack and fastest release. All other special features should be left at normal (eg hard knee, no sidechain, etc).

Now, lower the threshold until the compressor is reducing gain whenever the vocals are sounding, but not reducing gain when the vocals are not sounding (for example, in between lines or when the vocals takes a break to breathe).

You should hear the vocals being crushed pretty heavily. Now, how you reduce the crush depends on how you want the vocals to sound:

- Increasing the release time will make the compressor act slower, allowing each syllable or word to follow a more natural contour.
- Reducing the ratio will add dynamics back to the performance, increasing the difference between quiet words and loud words.
- Raising the threshold will allow the quieter syllables to pass uncompressed. The compressor will then only focus on the syllables that were sung louder.
- Increasing the attack time will make the compressor "lag" a bit when it's compressing the start of each phrase. This usually doesn't sound so good with vocals – it's more useful with drums and other percussive sounds.

That's the order in which I adjust my vocal compressors. It's very rare that I resort to advanced features like soft knee, side-chain EQ, etc.

This approach to compression also works well on other sounds, try it on bass, guitars, synths, effects, even drums.

-Kim.

2009/06/24 - Perception of time-speed

You may have noticed that the music listening experience is often not perceived as metric time. That is, *the speed of time* seems to vary throughout a piece of music. Some parts seem race by as if the clock were somehow accelerated, whereas other parts feels as if they last an eternity. This is the speed of time.

The word *perception* is important here because, while we cannot change the *actual* speed of time, we can (through music) change our perception of the speed of time.

Altering our perception of speed time can be done in many more ways than simply changing tempo. But first we must ask the question: *How do we percieve time?*

Imagine for a moment that you do not wear a watch, and you cannot see any clocks or other time-measuring devices. How do you measure time? Probably by remembering a number of events for a particular time period. This is not exact – "events" means anything that happens, and "time period" is your short-term memory. Both are variable. Consider two situations.

The first situation is that of sitting down, doing nothing for five minutes (or even better – watching the clock!). Time seems to crawl past.

The second situation is that of cleaning a messy desk in five minutes. Very busy, moving everything in its right place. The busy activity gives the appearance of time passing quicker.

In both cases, the elapsed time is five minutes, but the perception of time is different.

The same principal can be applied to music.

If you want to slow down your listeners perception of time, use less events and introduce less changes per time period.

If you want to speed up your listeners perception of time, use more events and introduce more changes per time period.

For example, we could focus on a section of music, and look at how many times the drum pattern changes, or how many chords there are, or how many notes (or note events) are in the melody.

It's important to note that while changing the number of events is useful, controlling the rate of change is most important. This is somewhat similar to the (more "classical") notion of "rate of presentation of material".

Another interesting way of looking at it is: Instead of measuring events+change per time period, look at *time period per change*. Approaching it from this angle, you might count the bars between each change, or look at the length of each section. A "faster" bit of music may have more sections, each shorter; whereas a "slower" bit of music mayb have fewer sections, each longer.

-Kim.

2009/06/25 - Buildups

Buildups often require particular attention when composing. A buildup section is one immediately leading up to a point of high energy. Commonly this is the climax of the song – the most important part of the song. The buildup is critical because it has to lead up to the climax in a way that maximises its effect. This is best done by enhancing the listener's sense of anticipation and expectation.

I've written about expectation in this post, but that alone is not enough. For a stronger effect, also consider sequences, precedent, linear movement and transition.

Sequences are repeated patterns in music. As explained in the post on expectation, sequences play an important role in setting the listener's expectations. Worth considering, though, is using sequences at different levels. For example, you could use repeated patterns within the buildup to create a cyclic effect.

Precedent applies this idea to a wider scope – whole sections. You could also make the buildup itself part of a larger sequence. By having a smaller buildup leading up to a smaller climax earlier in the song, you heighten the listener's expectation of a bigger climax at the end of a bigger buildup.

Linear movement is important in a buildup. Many song sections are static – they stay the same throughout the section (the same level of energy, the same types of sounds, the same density, etc). The buildup, however, works best if it is in motion. Typically, this works best if the buildup is gradually getting louder, more energetic and more complex. The buildup section might start quite subdued and understated, but at the end it might have enough energy to meet the climactic next section.

Transition is related to linear movement. It might help to think of the buildup as a transition section between the climax and whatever precedes the climax. Using the buildup as a transition section also helps glue the song together because it links the two adjacent sections (rather than sounding as a separate section on its own).

Another useful technique is to introduce a short "gap" in between the end of the buildup and the beginning of the climax. As explained in the last section of the post on expectation, this pushes back the climax and increases the sense of expectation and anticipation in the listener.

-Kim.

2009/06/25 - Volcano: Advanced Tactics

I've had another article published on ProRec, titled "Volcano: Advanced Tactics". It's a look into some of the more advanced signal processing techniques possible with FabFilter's Volcano plugin. Lots of audio examples too.

-Kim.

2009/06/26 - The vibe of a session

I recently came across this article, with this great paragraph:

> THE REALITY is that 90% of the time, the artist (and probably the producer) dont want to sit around and watch you turn knobs and swap mics until you get your idea of the most awesome sound. They want to record. Instead of the perception that you are doing your job to the fullest, the actual perception will often be "this engineer doesnt know what he's doing", and then before you've recorded a single note, everyone has already lost faith in your abilities, and the session vibe is blown. The most important thing in any session ever is the VIBE. A great vibe will usually translate to great feeling takes, which is a bit more important than the most amazing vocal sound. a bad vibe will equate to unusable takes, even is sonically they are wonderful. VIBE. believe it.

And this gem:

> The engineer is doing their job the best when they are transparent to the session. When nothing they do is slowing down the creative process.

My personal view has long been similar to this: Technology is best when it stays out of the way. As far as recording going, the engineer is part of the technology. The artist is in the studio to make music. Any time they spend *not* making music is time they spend *waiting* to make music.

Read the rest of the article. It's worth it.

-Kim.

2009/06/29 - Bit Depth

Occasionally there's a bit of confusion about bit depth, and about what the best bit depth to use in different situations is. In the digital world, there are three bit depths that we might have to deal with – 16 bit, 24 bit and 32 bit.

Bit depth determines the accuracy of low-level details in the audio. This includes the subtle details in the sound and the decay of notes or reverb tails.

16 bit

Digital audio at 16 bit is most commonly found in CDs. 16 bit audio allows the audio to have a dynamic range of roughly 96dB – that's the difference between the loudest possible sound and the quietest possible sound. This is fine for a final delivery medium – the vast majority of music has a dynamic range well within this limit.

On the other hand, 16 bit is not so good as a recording format. Often when recording audio, the nominal level has to be quite low so that accidental peaks don't distort (the distance between the nominal level and 0dBfs in a digital system is the amount of headroom). Because of this, low level signals (such as the decay of notes, or subtle details in the sound) may be recorded at a very low level below 0dBfs. When recording at 16 bit, any audio below 48dB (such as the decay of notes or subtle detail in the sound) is actually captured with less than 8 bits. This can give those low level signals a crunchy or distorted sound. This may be exacerbated in the mix by further processing such as compression and EQ.

24 bit

Professional analog-to-digital converters can capture low level details at higher resolution. This means that the low level signals can be captured accurately without having to record with less headroom. Recording at 24 bit allows the finest details to be saved. 24 bit recording provides a theoretical dynamic range of 144dB (compared to 96dB at 16 bit), but no analog-to-digital converter records with this much range (figures of around 110dB are typical). However, capturing at 24 bit is appropriate because computers are more efficient at handling data in 8 bit "chunks".

The problem with 24 bit audio is that it can be limiting when mixing. Mixing often involves summing a large number of tracks, each with several stages of processing. In this scenario, small errors can accumulate.

32 bit

Many software mixers convert audio to 32 bit internally during processing. This goes some way to reduce the effect of low level errors accumulating, and also has the added bonus of being able to have audio that excedes 0dBfs without clipping - as long as it happens internally to the software (that is, before it leaves the mix bus).

It is also sometimes worthwhile rendering audio at 32 bit. This would be a good idea if you intend to further process the audio. An example of this is if you render a mix to a stereo file, intending to import the stereo file into a mastering project. This means no resolution is lost between mixing and mastering. 32 bit audio is not suitable for recording or distribution.

It might be a good idea to start with an approach of recording at 24 bit, mixing at 32 bit and mastering to 16 bit for distribution.

-Kim.

2009/06/30 - Extending chords

Often, music that is based on scales and triads can sound harmonically simple. One way to make your sound more complex is to extend your chords.

An easy way to do this is to keep stacking thirds and fifths on top of the existing chords. For example, if you have a C-major chord the notes would be C, E, and G. To extend this, add B, then D, then F – all above the main triad. To keep it as clean as possible, don't let any notes be less than a minor third apart – add the additional notes in the octave above.

Another example – you're working in F#-minor. Your triad is F#, A, C#. You could extend it by adding E, then G#, then B.

As easy way to think about it is to just keep alternating between major thirds and minor thirds. In the C-major example, C to E is a major third, E to G is a minor third, then G to B is a major third, then B to D is a minor third, etc, etc.

At first it may sound a little strange, but you'll get used to it.

This method is useful (I use it myself) because you still retain your tonal centre (your bassline still makes sense), and the additional complexity is added gradually – meaning you can control how complex you want your chord to be. If you just want a bit of added complexity, just add the next note above. If you want more, add another note. Or another. Or another.

You could extend this idea by having a different instrument play the extentions. For example, you could have your favourite thick pad playing your triad, then have the extentions played by a thin airy pad in the background. This gives you more ways to balance the simple with the complex. Some synth pad presets in workstation keyboards do a similar thing automatically – each note has the main sound on the note and a lighter note a fifth above.

-Kim.

2009/07/01 - New Robot Child songs up

Just a quick post to say I've uploaded two more Robot Child songs – One Final Scream and It Won't Last. Both are a bit rockier than their previous songs.

I'm not in the band, but I've mixed all five published Robot Child songs. It's not a bad sound for what is essentially a single take in a rehearsal studio (plus a couple of overdubs). I just wish drums didn't sound so much like buckets and tin cans.

-Kim.

2009/07/01 - Sample Rate

As with bit depths, there are several different samplerates used for digital audio. While bit depth determines the accuracy of low level details, samplerate determines the accuracy of high frequency details.

Samplerate is actually the rate at which the digital audio is being processed, and where there are multiple files or audio streams being processed simultaneous (such as in a DAW or digial mixer) all the audio streams need to be at the same samplerate. For this reason, it's best to choose a samplerate before recording begins, and avoid changing mid-project.

44.1 kHz

This is by far the most common samplerate, as it is the samplerate used by audio CDs. Roughly speaking, the highest frequency that can be represented is exactly half the samplerate – so at 44.1kHz, the highest possible frequency is 22.05kHz. Seeing as most people cannot hear above 20kHz, this would seem like a good samplerate choice. The problem with this, however, is that the *accuracy* of those high freqeuencies is quite poor. The closer you get to the highest possible frequency, the worse the accuracy gets. As a result, the inaccuracies can sometimes be heard well below the highest frequency. For this reason, many plugins *oversample* their critical processing components – meaning the audio is internally converted to a higher samplerate so the highest frequencies can be processed with better accuracy.

48 kHz

This samplerate effectively has the same limitations as 44.1 kHz, except that it's more commonly used for film and other visual media. This is because it syncronises better with visual frame rates (44.1 kHz doesn't divide evenly into 24 frames).

96 kHz, 192kHz

Most modern professional digital audio systems can operate at higher frequencies than 44.1 kHz or 48 kHz. This allows audio to be captured and processed with much higher accuracy, especially at the highest frequencies. This usually results in a more open, natual sound. The trade-off is that much more processing power is required. Working at 96 kHz will half your capability compared to working at 48 kHz. That includes disk space (recording time), disk throughput (number of simultaneous tracks) and CPU/DSP power (number of compressors/EQs/effects). Working at 192 kHz cuts your capabilities in half again. Whether this trade-off is worthwhile for you depends on the kind of work you're doing and your style of working. If you want to record the clearest, purest sound with a minimum of processing, high samplerates might be appropriate. On the other hand, if you want to do a lot of processing (especially if you regularly push your equipment to the limit) then you might prefer the higher capabilities of working at a regular samplerate.

Recording at these high samplerates also has an advantage for sound design and special effects. Because there is so much more high frequency detail captured, slowing down a high samplerate recording results in a much clearer sound than slowing down a sound recorded at 44.1 kHz or 48 kHz.

If you work at 96 kHz or 192 kHz, you might need to convert back down to 44.1 kHz or 48 kHz when preparing audio for distribution.

-Kim.

2009/07/06 - Writer's block

We've probably all experienced writer's block at some stage. Sitting in front of the computer, keyboard or guitar, wishing an idea or inspiration would come. Instead, no music comes. It brings down our self-esteem, which further inhibits creativity. It's a horrible cycle.

I've seen a a lot of advice about how to deal with this kind of situation, and it seems most of it is a variation on the theme of "doing something random". That is, bringing an unexpected or unfamiliar element into the music. This ranges from using a new instrument, to limiting the song within certain parameters (such as time, key, instrumentation, style, etc).

I disagree with this advice. In my experience, this is not the path to productivity.

For me, the first step is to get some distance. Get out of the studio. By physically removing myself, I walk away from the (self-imposed) burden of having to force creativity. If the weather's good, I like to take a walk. The fresh air combines with the physical activity, stimulating bloodflow and brain activity. I use this time to reflect on my current projects. What progress has been made on each, what further work needs to be done. I remind myself of the overarching creative direction for each project – what is the style of music? What is the style of working? What are the expected outcomes?

If you don't have any projects, this is the time to begin planning one. Choose to embark on either an album-length (12 songs) project or an EP-length project (6 songs). Think about some music you've been listening to, and consider some aspects of the music that you find interesting or inspiring. Think about combining different musical approaches or trying new musical approaches. Think about a unifying theme.

If you already have a project that you're stuck on, think about the bigger picture. What are you trying to achieve with the project? If you're stuck for creative direction, think about aesthetic – colour and texture. If you're stuck for lyrical ideas, think about the key message of the song. If you're stuck for ideas for a new song, think about music you've always wanted to make.

Either way, don't return to the studio until you have a plan. Don't return until you already know what you're going to do when you sit down in the big chair. Make sure that when you return you know exactly what you're going to do.

-Kim.

2009/07/07 - Limiting vs Clipping

Limiting is an extreme approach to compression. Where compression reduces the degree by which sounds can go louder than the threshold, limiting is designed to stop sounds from being any louder than the threshold at all. Limiters usually have simpler controls to compressors, but are functionally similar to compressors with high ratio and fast attack.

Limiters are useful for reducing peak level (the level that machines "hear") of a sound without affecting the average level (the level that humans hear). This reduces the headroom that the sound needs so that it can be made louder without distorting.

The downside the using limiters is that they reduce the level in the same way that compressors do – by applying gain. That is, they *turn down the volume*. Ideally, a limiter does this fast enough that the transient peaks are reduced but the steady-state sound is not audibly affected. In many cases though, the gain reduction is audible. It can give the sound a soft, wooly character, or even a random tremolo effect if the threshold is too low. Often using a limiter reduces the power and impact of a sound. This effect is sometimes hidden because the limiter also increases the overall volume, making it more difficult to notice that the character of the sound is changed.

An alternative to limiting is clipping. Yes, this is the same clipping that usually engineers try to avoid when recording and processing audio. Most of the time clipping is undesirable, but in some situations it can be used deliberately and beneficially.

While limiting reduces the level of the transient peaks by turning down the volume (applying negative gain), clipping reduces the level of transient peaks by distorting them. If you view the waveform of a clipped sound, you'll see that the waveform looks like it's had the tops and bottoms chopped off. This is distortion! However, if only the transient peaks are clipped, the clipping only occurs for a very short period of time, and is not very noticeable.

When this happens, the excess level in the transient peaks is transformed into upper harmonics. That is, the transients become *noisier and dirtier*. For some kinds of music, this can be a desirable alternative to reducing gain. The power and impact of the sound is often retained (or even enhanced!), but at the expense of fidelity.

-Kim.

2009/07/08 - Gain Staging

A "gain stage" is any point in the signal path where gain is applied – where volume can be changed. Gain can be positive (makes the sound louder), negative (makes the sound quieter), or unity (doesn't change the volume – but it's still a gain stage!). "Gain staging" is the awareness that there are all these gain stages, and it's important to carefully adjust each one so that each processing stage is operating optimally. This means balancing headroom and noise floor to keep the audio as clean as possible.

The noise floor of an audio system is the level at which the background noise (hiss, etc) is. This is not the hiss in the recording, but the background noise inherent to the system itself. Generally, it's best to stay as far away from the noise floor as practical. In analogue systems, the noise floor is hiss or hum caused by the electrical components. In digital systems, the noise floor is crunchy quantisation noise caused by a lack of digital resolution. In modern digital systems, noise floor is not a big concern. Most professional analogue-to-digital converters (ADCs) have a noise floor below 100dBfs.

The headroom of an audio system is the amount of room (in decibels) between the 'nominal' level and the saturation level. The nominal level is level at which the audio spends most of its time. There is some flexibility in deciding what the nominal level should be. A low nominal level will give you lots of headroom, but a higher noise floor. A high nominal level will give you less headroom but a lower noise floor. The less headroom you have, the more saturation/clipping you'll get, and the more compression and limiting you'll need to keep the sound clean.

Noise floor is less of an issue in professional digital systems (especially all-software systems such as DAWs), but headroom is still critically important – even more so in today's loudness war. If you don't give yourself enough headroom early in the signal path, you'll find yourself hampered by your need to reduce dynamics for technical reasons instead of focussing on sound.

-Kim.

2009/07/09 - Drum programming – Expectation and Excitement

This post is about drum programming, but these prinicpals apply to all aspects of rhythmic composition (including basslines, melody, etc). Additionally, these principals are applicable to drum patterns of arbitrary (any) complexity, but for simplicity we will be primarily concerning ourselves with the four-on-the-floor kick drum pattern often heard in popular club music.

A little more on drum patterns of arbitrary complexity. The important thing to remember here is that everything is relative to the normal. That may sound somewhat obvious, but I'll explain further. A static (looped) drum pattern can only be interesting for a limited amount of time. After hearing the repetition several times, we (the listener) know what to expect. However complex this drum pattern is, repetition makes it the normal. If, after a few iterations, we change the loop, we (the listener) will be surprised in some way… but if this variation is then looped, it becomes the new pattern – it becomes the new normal upon which we build our new expectations.

So basically, we will discuss two patterns – the normal, and the variation. The normal is what has been repeated, and what the listener expects. The variation is a new loop that is very similar to the normal, but different enough to surprise the listener.

Expectation

We can create expectation by removing notes from our variation. The sense of expectation is created because the listener expects (from the normal) a certain note to exist, but it does not. You might say the listener "wants" something to be there, but it is not, so the listener is kept "wanting".

For example:

```
normal  variation
1 2 3 4 1 2 3 4
```

```
* * * *,_ * * *
```

or:

```
normal  variation
1 2 3 4 1 2 3 4
* * * *,* _ * *
```

If these examples don't make immediate sense, imagine that the first four beats ("normal") are repeated several times before the next four beats ("variation") are played. If it still doesn't work, turn up the volume.

We can take this idea further if we look at the concept that rhythmic patterns usually have a hierarchy of "strong" and "weak" beats. Beats 1 and 3 are usually the strongest (at least in modern popular music), then 2 and 4 are weaker. The rhythmic positions "between" the beats are weaker still, etc.

Coming back to our discussion of expectation, we might observe that removing notes on strong beats tends to (comparatively) emphasise the weaker beats. This enhances the expectation because it creates more tension – the pattern wants to "resolve". Some examples:

```
normal  variation
1 2 3 4 1 2 3 4
* * * *,_ * * *
```

or:

```
normal  variation
1 2 3 4 1 2 3 4
* * * *,* * _ *
```

Or deviating from four-on-the-floor:

Code:

```
normal   variation
1 2 3 4 ,1 2 3 4
*  ** **,*  *_ **
```

Again, imagine that the "normal" pattern is repeated several times before the "variation" is played.

Alternatively, we could remove notes on weak beats. Comparatively, this tends to emphasise the strong beats, and thus doesn't create as much tension, and doesn't reinforce the expectation. Examples:

```
normal  variation
1 2 3 4 1 2 3 4
* * * *,* _ * *
```

or:

```
normal  variation
1 2 3 4 1 2 3 4
* * * *,* * * _
```

You'll observe that creating expectation by removing weak beats is generally not as effective as removing strong beats. In fact, removing weak beats is more effective at thinning the texture, and this is what is more likely to be percieved.

Excitement

The flip side of expectation is excitement. Excitement in rhythmic patterns can be created by adding notes in. Example:

```
normal  variation
1 2 3 4 1 2 3 4
* * * *,* * ***
normal  variation
1 2 3 4 1 2 3 4
* * * *,* * * **
```

The reason this creates excitement is that the listener is (from the "normal") expecting nothing, and in it's place they get something. Onother way of looking at it is that the listener hears something before they expect it. We can extend this technique by observing that when an added note is close to an existing note, the added note can associate with its neighbour.

To show an example of this, we'll need to double our resolution:

```
normal         variation
1   2   3   4  1   2   3   4
*   *   *   * ,*  **   *   **
```

When an added note associates with an existing one, it can have an effect of reinforcing the existing note. Additionally, the position of the added note (whether it comes before or after the existing note) can also have an effect on how the existing note is reinforced.

If the added note comes before the existing note that it associates with (variation, beat 2), then it tends to create the illusion that the existing note comes earlier than expected. This creates excitement for precisely that reason – the note comes earlier than the listener expects.

If the added note comes after the existing note that it associates with (variation, beat 4), the effect is that of strengthening the existing note, or elongating it (making it longer).

-Kim.

2009/07/13 - Sections of variable length

Often I've found that using sections of "metric" lengths (four bars, eight bars, sixteen bars) can often give a piece a very rigid, predictable pace. No matter how exciting or interesting the actual musical material is, sections of metric length can really weigh a piece down.

This is because the listener knows (or can guess fairly accurately) when each change will occur. In her/his mind, the listener has heard a sigificant amount of the piece *before it's actually been played*.

Let me give you an example. Let's say that so far, every section has been sixteen bars long, and it's very obvious whether each section is static or transitional. Within a few bars of hearing a particular section, the listener already knows what the rest of the section sounds like – sometimes to the point of *not actually having to hear the remainder of the section*. This is the point at which the listener becomes distracted, starting to talk, or getting bored.

A particularly effective way to reduce this effect is to use variable section lengths. Instead of making each section a "metric" length (four bars, eight bars, sixteen bars, etc), the idea is to make them "odd" lengths. This has two implications:

- The listener will not be quite so sure how long each section will be. In fact, (if done well) sections will often end/change earlier than expected or later than expected. This can be taken advantage of to highten expectation and excitement.
- The internal structure of each section will be more "fluid": In sections of "metric" length,

> we tend to break them up into smaller bits of even length. For example, if we have a section of sixteen bars, we might very easily put in eight chord changes, one every two bars; or four chord changes, one every four bars. If we have a section with an "odd" length, it forces us to be more creative with the internal structure. For example, if we have a section that is thirteen bars long, we might split it into three groups of four bars, plus one; or four groups of three bars, plus one; or three groups of three bars, plus four bars… or anything else.

How you come up with the lengths is up to you. I composed a piece a several years ago where each section length was a Fibonacci number – the sections were all lengths like 5, 13, 21, 34, etc.

Another piece I composed had section lengths chosen by rolling dice.

Of course, it doesn't have to be random. You might choose prime numbers, or the date of every Monday in the year, or anything else. You could even choose the lengths as you compose the piece, depending on the flux in the piece.

It's really just about making the sections have lengths which aren't even multiples of four or eight.

-Kim.

2009/07/15 - Composing for Kick Drums 1

Kick drums. Where would we be without them? They are the foundation of the rhythm section. In most dance music, the kick drives the rhythm and groove of the entire song. Even in other genres, the kick drum provides a grounding. It marks the most important beats in the rhythm pattern, it helps us understand the rhythms of the rest of the drum kit, other percussion, and even other instruments.

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With such an important role, have you ever stopped to think about how you use kick drums? Do you place them on every beat, four-on-the-floor style? Do you arrange them off-beat, with a more funky style? Do you lay them thick with five or six every bar, or is one or two enough for you?

Even more importantly, how do these different approaches differ? What effect do they have on your music?

Four On The Floor

This is the simplest kick drum rhythm, with one kick on each beat. It's also one of the most popular. This is particularly useful in dance music or driving rock because it's regular (meaning it's predictable, comfortable, and easy to dance to). It's also quite energetic because it emphasises every beat. Even though a bar may consist of four beats, it almost feels like each bar is one beat long. The shorter cycle length (the pattern repeats every bar) make the pace feel quick.

This approach is most useful where the creative direction for a piece (or section) is "stable, yet exciting". A prime example of this would be a climax final chorus of a song, or a section of maximum impact in a dance track. The drawback, of course, is that it's plain and not very interesting on its own. The four-on-the-floor should not be the source of the excitement – merely underpinning it and reinforcing it.

First and Third

Using the kick drum on the first and third beats is a sparser variation of the four-on-the-floor approach. By playing the kick only on the first and third beat, a lot of room is left for the snare – either for a big second and fourth, or for a busier, funker snare pattern. This is useful if you are aiming for a more top-heavy drum kit rhythm, or if you want a sparser drum rhythm to leave room for additional percussion or other instruments like vocals or bass.

The drawback is that this sparser approach does not reinforce excitement as much as a busier four-on-the-floor kick drum pattern. It can still be effective, however, because it is just as stable and predictable. It can be a good alternative to four-on-the-floor if the song needs space to breathe or otherwise doesn't need the relentless kick of more upbeat music.

More coming…

-Kim.

2009/07/16 - Composing for Kick Drums 2

Sparser Kick Drums

In general, sparser kick drum patterns will be less energetic. As with the First and Third pattern, a sparse approach is generally useful for leaving space for other instruments. Taking this approach, the kick typically only emphasises the first beat of the bar, and sometimes a secondary beat (secondary in importance – not necessarily the third beat). It can also be a very effective way of giving a section or whole song a slower *pace* without slowing the *tempo*. With a busier snare or other percussion, a *very* sparse kick can be very ear-catching because there may be implied beats or expected beats that aren't there.

Denser Kick Drums

In general, denser kick drum patterns will be more energetic. In the context of the whole drum kit and other percussion, a lot more emphasis will be placed on the kick. This can be a problem if the snare or other percussion is also big and/or busy. For a balanced approach, it might be better to combine a dense kick drum pattern with sparser, smaller snare and other percussion. If If the kick drum is particularly prominent (such as in many dance genres), the other instruments may need to be thinner than usual to accommodate as well. On the other hand, a dense kick drum pattern is a good way to emphasise a heavy sound with a strong rhythmic focus. The best example of this is heavy metal, where there are extended passages with double-kicks (constant kick drums on 8th notes or even 16th notes!).

Off-Beats

Most kick drums notes fall on the beat – meaning they are played on quarter notes (also called crotchets). The two patterns discussed last time (Four-on-the-floor, and First and Third) have kick drums played *only* on the quarter notes. Sometimes, however, it sounds good to play the kick drum on an "off-beat" – in between the quarter notes. Notes played on off-beats are less stable and (mostly) less predictable than notes played on the beat. If you have a lot of notes played off-beat, and not as many notes played on the beat, the whole patteren will feel more unstable, more unbalanced, and more unpredictable.

With a careful balance of on-beats and off-beats, funkier patterns are possible. These balance stability with instability on a moment-by-moment basis. Typically there will be a kick on the first beat (the *downbeat)* of every bar (or only every second bar!) to ground the listener and begin from a point of stability. In the middle of the bar, however, the kick may be played at various points on or off the beat. This creates a constant push/pull between stability and instability, and can make a pattern much more exciting and interesting to listen to. The effect is heightened when the kick only plays on the downbeat every second bar – so the other bars don't even start with a kick.

Swing

Some very interesting things can happen when introducing swing to kick drum patterns[1]. To hear the swing on a kick drum, it already has to be playing off-beat (that's how swing works – by delaying the off-beats). When a kick drum is swung, two things happen:

1. The kick plays later than expected – meaning the listener is kept waiting in expectation for a (very) brief moment. This contributes to the push/pull of stability and instability, which is also related to expectation.
2. The kick aligns closer to the coming beat – meaning it emphasises the anticipation felt by the listener.

This further emphasises the subtle push/pull of the kick drum pattern.

More coming…

-Kim.

[1] Sometimes I rhyme, but not all the time.

2009/07/17 - Composing for Kick Drums 3

Variation

Like for any other part, adding variation to the kick drum pattern adds interest and scope. Generally speaking, there are two kinds of variation – changing the timing of notes (keeping the same density) and adding/removing notes (changing the density). These variations are most effective when a regular pattern has been established (repeating for several bars) before presenting the variation bar. This frames the variation pattern in the context of the regular pattern. This effect does not last long though – if the variation bar is repeated several times, it becomes the new regular pattern. The effect of any change (such as excitement or anticipation) fades.

Pulling Notes Forward

This is changing the timing of a kick drum note so it plays earlier than expected. For example, if the regular kick drum pattern is "First and Third" (kick drum plays on the first and third beats of the bar), a variation might be to pull the second kick forward from the third beat to halfway between the second and thid beats. This will add a sense of excitement for the listener, as they hear the kick drum play earlier than expected.

Pushing Notes Back

The opposite of pulling notes forward is to push them back. This is where kick drum notes are played later than expected. Using the same example ("First and Third"), a variation might be to push the second kick back from the third beat to halfway between the third and fourth beats. This will add a sense of heightened expectation and anticipation for the listener because the kick drum doesn't sound when expected, but the effect is tempered by eventually providing the kick a little later.

Adding Notes

Adding notes as a variation is a more effective case of pulling notes forward. The added kicks are heard as occurring earlier than expected, but the "original" kick is also heard. The added kick also increases the density of the pattern, which also adds excitement.

Missing Notes

Missing notes is an extreme case of pushing notes back. Instead of simply changing the timing of a kick so it is heard later than expected, the kick is removed altogether. This results in a sense of anticipation that isn't fully resolved – it is only partially resolved when the *next* kick hits.

-Kim.

2009/07/20 - Normalising

Normalisation is a process that changes the volume of a piece of audio. It does this by first analysing the audio, looking for the highest peak. Then an amount of gain is applied to the entire section of audio, so that the highest peak is at 0dBfs. Because of the need to analyse the audio before applying gain, normalisation is an offline process – meaning it can't be applied in realtime (as a plugin, for example). Also, because static gain is applied, the dynamics of the audio do not change. It's exactly the same as adjusting the fader on an audio channel, except that there is a pre-calculation to determine how much to adjust it.

There are two problems with normalising:

1. You don't know or control how much gain is being applied. That's because the amount of gain is determined by analysing the audio.
2. The amount of gain being applied has nothing to do with how loud the audio is (as we perceieve it). That's because the amount of gain is calculated from the peak level of the audio – not the RMS or average level (see here for more details about peak vs RMS).

Normalising audio ONLY makes sense if:

1. Your audio started higher than 16 bits; AND
2. You're about to quantise to 16 bits (or similar) directly after normalisation; AND
3. You don't care what the average (RMS) level of the audio is after quantisation.

In other words, this is a process that makes sense where there are a series of offline gain stages, and somewhere in the MIDDLE the audio is being quantised to 16 bits (but subsequent processing is at a higher bit depth). The uncontrolled amount of gain is not a problem if later gain stages will also be applied.

In these situations, normalising is a useful way to maximise the dynamic range of a low-resolution digital system. This is because the audio is made as loud as possible before quantising so that the higher noise floor (caused by the low resolution) is as low as possible relative to the audio. An even lower relative noise floor is possible by using dynamic processing (such as compression or limiting), but normalisation is the best solution that doesn't affect the original dynamics of the audio.

On the other hand, if your task doesn't meet all three criteria, then there are more appropriate processes than normalisation.

-Kim.

2009/07/21 - Beginning, middle, end

Normally when we compose a piece of music, we are working on it in a non-linear fashion. That means we can work a little on the start, then work on the end, then maybe add a new section in the middle, whatever. Also, our perception of the piece is non-linear – being so intimately involved with the piece (and its construction), we usually know the entire piece by memory. That gives us (the composers) the ability to compose parts of a piece in the context of the rest of the piece.

Your listener, however, will have a very different experience of the music. As an artform, music is particularly interesting because it exists in time. You listener will listen to your piece by starting at the start, listening through each moment once, and stopping at the end.

Consider that the listener will also have no (or at least, very little) knowledge of the piece before listening. S/he will begin the listening experience knowing nothing, and gradually (and linearly) learn more about they piece as it is experienced. I like to think of this as an "unravelling" or "unfolding" of music – as the listener experiences the piece, it is being revealed, opened up.

This observation has interesting implications for different sections of a piece.

The beginning is significant because it "introduces" the language of the music to the listener. When you listen to a piece of music, the beginning is the first thing you hear – and thus, it is what influences the expectations that you have for the rest of the piece. It is what sets the context for the remainder of the listening session. When composing the beginning of a piece, consider that this is the first thing your listener will hear.

The ending is (in this respect) the complete opposite – the listener hears it in the context of the entire piece. By the time the listener gets near the end of a piece, s/he has travelled through the "journey" of the music, and (hopefully) understands the language[1] of the music. When composing the ending, consider that the listener hears this after hearing the entire piece through once.

The middle of a piece is also interesting, because this is (usually) where the "scene has been set" – the listener has some idea about what the laguage of the music is, and what to expect for the rest of the piece. Most well-written pieces use a/the middle section to develop and enrich the listeners understanding and experience of the world you (as the composer) have created.

All this, of course, doesn't mean that this is the way it has to be, or that this is the (only) way to compose "good music". As the composer, you are free to subvert the rules or discard them completely. However, understanding how a listener listens to a piece will (hopefully) help you make better informed decisions during the composition process.

-Kim.

2009/07/22 - Transition sections that are too long…

If you have a transitional state between two sections that have similar rhythm, pace, tonality, register, texture, etc; it doesn't take much time to move between them. However, if you try to move between two very different sections, more time will be required (for the same rate of change). The time required to move from one state to the other is somewhat proportional to the "distance" (or difference) between the two states.

Sometimes it's important to have a long transition in order to put some distance between two sections.

The problem is that a linear transition gets boring very quickly. Alinear transition is one that just goes straight from A to B. For example, if you have a dark and resonant synthbass sound in section A and a bright and harsh synthbass sound in section B, then a linear transition would be one that (among other things) just gradually opens the filter, like a straight line – slow and predictable.

And that's exactly why a linear transition is boring – because it's predictable.

If reducing the length of the transition is not an acceptable solution, another alternative is to break it up.

- **Change the "curve" of the transition** – Rather than move predictably from A to B, perhaps start the transition with a very low rate of change, and gradually increase (the rate of change)… This would have an effect of the transition section intially not sounding like a transition – but more static (unchanging). Depending on the context, this can either create tension (possibly good) or boredom (probably not good). Slowly we hear more an more changes happening (the changes speed up), "climaxing" with a very (or more) dramatic change at the end – just as the next section begins.
- **Put breakpoints in the transition** – Rather than simply starting at "State A" and moving towards "State B", you could add animation between the two points. For example, the transition moves from A, but when it gets about halfway between A and B, it goes back to A again. Then we start moving towards B again, but this time get about 75% before moving back again. and maybe a third time we finally reach B. This can be particularly interesting if the listener already has some knowledge of where the transition is going (s/he has already experience B in some form). another interesting effect of this is that it can distort the listener's sense of time as well, but this is more difficult to control and excecute well. This technique could also be extended by including "fake" transitions to a different state (not A or B) – this will also manipulate the listener's expectations.
- **Insert contrasting sections in the transition** – Four bars of transition, then four bars of something completely different, then the four next bars of the transition, then another four bars of something completely different, etc etc etc. Of course you can vary the length of each "mini-section", you could have each contrasting section be something relevent from another part of the piece. You could effectively interleave two different whole sections – even have two transitions coming from different states but both moving to the same final state.

Of course, these ideas aren't restricted to transition sections – you can use variations of these ideas to add interest to any long section. You could even apply them to only some elements of the arrangement (for example, make the synth parts go through a transition, but keep the drums and bass consistent).

-Kim.

2009/07/23 - Nested Structures

Nested structures are quite simple to understand, but can add new levels of order and structure to your music.

If we start with two basic structures:
Binary: **A B**
Ternary: **A B A**

Nested structures refers to the idea that each of the structure "elements" (A, B, whatever) can actually be (or have) a whole structure in itself. This can be the basis for develping more complex structures from simple ones. For example:

We could choose ternary for our overall structure (**ABA**). But if we split it up further – replace **A** with the binary structure **ab**, and replace **B** with the ternary structure **cdc**, then we end up with the overall structure **ab.cdc.ab**. Read through this a few times if you didn't quite get it.

Now think about taking that another layer deeper. You can keep nesting structures until you get down to individual phrases, gestures, motifs, even notes!

Also, consider that there are many more possibilities for "basic" (or primative) structures. As well as binary and ternary, there's also rondo (ABACADA – commonly chorus,verse,chorus,verse,chorus,etc), sonata (A B <development> A B')

Usually, there's two ways to approach this: *top-down* and *bottom=up*. A top-down approach would be very similar to my example above – start with an overall structure, and then split it up into smaller and smaller pieces, stopping when you feel that you can easily populate a single piece. This is a "divide and conquer" approach.

A bottom-up approach would be the exact opposite – start with several very small pieces, then arrange them into larger and larger structures. This is very easy to do in a sequencer, where you can develop a few one or two bar sections, then copy and paste them in various orders and configurations.

Personally, I usually use a combination of the two. I build the piece bottom-up, but when I'm doing it I have a mental "plan" of how I want the entire piece to turn out.
By approaching composition in this way, you can create a piece with a very high level of coherence and order. Each section will fit exactly in its place, and repeated sections can give a certain unity – without having to resort to a simple verse-chorus-verse-chorus (or similar) structure. It's also an easy way to add complexity by using relatively simple ideas.

Of course, the fun begins when you combine nested structures with techniques for subverting nested structures. Build your piece as usual, but add interest with variations, interruptions, twists and turns, bizzare trips to strange places. These kind of subversions are usually much more effective when you start with something with very high coherence and order.

-Kim.

2009/07/27 - What is saturation?

Saturation used to be something that happened in the analogue world. Typically, this is when a gain stage is overloaded – the signal level exceeds the available headroom. When this happens, the signal is saturated.

Basically, the sound gets distorted because you turned it up too high.

The result of this is that the parts of the signal that were going to exceed the available headroom are *waveshaped*. If you were look at the waveform of a saturated signal, you'd see that the loudest parts of the sound have been clamped down – they're quieter than they should be. This is similar to what a compressor does, except that saturation affects the shape of the waveform itself – not necessarily the perceived volume level (as we hear it). By changing the shape of the wave, the sound changes too.

Transients

Saturation reduces the level of transient peaks by distorting them. However, because the transient peaks are very short, the disortion is often not obvious. The excess level in the transient peaks is transformed into upper harmonics. That is, the transients become *noisier and dirtier*. For some kinds of music, this can be a desirable alternative to reducing gain using a compressor or limiter. The power and impact of the sound is often retained (or even enhanced!), but at the expense of fidelity.

Steady-state

Saturation of steady-state signals is often more noticeable because the audio is constantly being saturated. This usually causes the sound to be brighter, as upper harmonics are being created. Too much saturation will make the audio sound lo-fi or outright distorted. Used subtly though, saturation can make audio sound more exciting, or even aggressive. In a dense mix, individual tracks will sound less distorted than when listening to those tracks on their own (in solo).

-Kim.

2009/07/28 - Recording vocals

My process for recording vocals for songs is as follows:

0) (Write the song)

1) Record a guide track. This is usually a skeletal piano part that simply plays the chords for the song. And simple means simple – it's usually block chords, one per crotchet (1/4 note). Sometimes there's a particular riff or groove that's integral to the song, and this forms part of the guide track. The guide track rarely makes it to the final mix – its sole purpose is to provide the vocalist with something to inform their pitch and timing. Sometimes the guide track doesn't represent actual structure of the song – I might record the sections separately and arrange them later while I'm recording the other instruments.

2) Record a scratch vocal part. This is performed to the guide track. Usually I only record one or two takes of the lead vocal melody, and harmonies or backing vocals only if they're particularly important. This scratch vocal recording only needs to serve as my own guide for writing for and recording the other instrumental parts. As such, it doesn't need to be a great performance – it might even have some mistakes in it! The bulk of the time in this session is not spent coaching the vocalist – it's spent working on song structure and melody before the recording takes place.

3) Record other instruments. This is done using the scratch vocal as a guide for song structure and mix placement. For vocal music it's important to start with the lead vocal and build everything else around that. This is important not only for arrangement, but also for instrument voicing (which notes, how high, etc) and mixing. Producing a backing track without having a vocal to work with will easily result in an instrumental song that sounds great on its own, but will struggle the accommodate the vocal once its added.

4) Record backing vocals. Once the arrangement is worked out and most of the other instruments are recorded, it's time to record the final vocal parts. It's important to wait until the track is almost finished, so that the vibe and energy of the music can influence the vocal performance. That way the vocalist can deliver a performance that best suits the song.

I usually have the vocalist record the backing vocal parts first. This is because they're not as critical as the lead vocal. This allows the vocalist to warm up and familiarise himself/herself with the song and the studio. It gives us both a change to fine-tune the headphone levels and monitoring (usually some compression and treble lift) whilst still remaining productive. It also gives me a chance to learn how to work with the vocalist to get the best results. Some will respond well to relentless pace, some resond better to a gentler approach. Some nail it in the first couple of takes, others need six or eight takes to get it. Some respond well to micro-advice ("That fourth syllable is dragging again"), some need more general encouragement ("That was great, now do it with more energy"). It also helps me learn about the voice. How hard can it be driven? How soon until it needs a rest? Where is the sweet spot between warming up and tiring out? Are there any difficult transitions between chest voice and head voice? These are all important issues to be aware of.

5) Record the final lead vocal. This is critical. The lead vocal is the most important part of the song. As if that wasn't enough pressure, the vocalist only has a short period of time in the sweet spot at which you'll get the best performance. You've got to know where this sweet spot is, because going much past it will give you worse takes, and demoralise the vocalist. Coming back another day often doesn't give you a better performance either (unless you screwed up and scheduled the session when the vocalist is hung over or something).

My approach to recording lead vocals is the result of many years of working with vocalists. Everyone will have their own methods, but this works for me:

5a) Run through the song once for practice. Often I see advice to record the practice takes. I don't record them, because I know I'll get better later on. I try to minimise the number of takes I record, because trawling through them later is a chore, and often doesn't actually get significantly better performances. Some vocalists don't even need the practice take.

5b) Record one take of the whole song. This is the basic lead vocal. I only record one take of this, unless there were any mistakes (in which case I delete the take and do it again). This is usually a good fallback for syllables to comp[1] in.

5c) Record two takes of each section, in reverse order. After the first take, we record section by section, in reverse order. So, we'll start with the coda, then the final chorus, then the bridge, etc, finishing with the first verse or introduction. Each section is recorded twice on loop. I'll only go more than twice if there were any mistakes. Recording each section in reverse order frees the vocalist from thinking about the song structure, and instead focusses her/him in the moment – the section being recorded. Recording each section twice on loop further enhances this focus. This is sometimes the point at which the singer is in the sweet spot, and where you'll get a good balance between emotional performance and technical correctness. I record only two takes during this phase in order to keep the momentum up (which keeps the singer motivated and interested). By this stage, more than two takes rarely results in a better performance.

5d) Record one take of the whole song. At this point I have lead vocal that's three takes deep, and is usually of a consistently high quality. I could use this material to put together a decent composite. However, I ask the singer to do one more take – the whole song through. This time, however, I instruct the singer to focus on delivering an emotional performance at the expense of technical correctness, perhaps pushing it a bit more than usual. I might even bump the volume in the headphones by a decibel or two to assist. For some singers, this falls in the sweet spot and results in a great performance. For others, it's too much, so the take becomes a source for the occasional 'emotional syllable'. If this last take is not suitable, I'll usually use one of the 'reverse order' takes as the base for the comp. Most times, however, it is this finial take that forms the base from the comp.

-Kim.

[1] Comp = Composite. The process of assembling a performance from several takes.

2009/07/29 - Soft-knee compression

So, you think you're pretty familiar with compressors. You know how they work, what they do. You know what the basic controls do – attack, release, ratio and threshold. But maybe you're not sure about 'knee'. Something about being soft? What is knee, and – more importantly – what is it useful for?

Normal compressors behave depending on the level of the audio. When the audio is below the threshold, nothing happens. When the audio is above the threshold, the level is reduced by an amount related to the compressor ratio. This is what's called a 'hard knee' – there is an instant transition between unity ratio (1:1) and maximum ratio (whatever the ratio knob is set to). Soft knee, by comparison, causes there to be a gradual transition from unity ratio to maximum ratio. This means that some audio that is only just below the threshold may be compressed, but at a lower ratio than audio above the threshold.

It's useful to think about soft knee as a kind of variable ratio. It's especially useful when you want low-level signals to be compressed a little, but high-level signals to be compressed more aggressively. A good example of this is a vocal performance where most of the audio is fairly even but there are some syllables that are quite loud. If you use hard knee compression, you'll have three options:

1. High threshold, high ratio. This will control the peaks, but the rest of the performance will be uncompressed. You might need a second compressor to apply some more gentle leveling.
2. Low threshold, high ratio. This will bring all the levels into check, but may sound too aggressive or choked – especially if you're aiming for a more natural sound.
3. Low threshold, low ratio. This will give you the gentle levelling, but the loud syllables will still pop out due to the low ratio.

Using a soft knee, you can apply some gentle levelling compression, while still controlling the loudest peaks. Set the ratio to the maximum you'll need, and gradually lower the threshold until the lower-level audio is being gently compressed. That way you'll have the gentle levelling with a low ratio, but the louder parts will be compressed at a higher ratio. It's a little like having two compressors in series (one reducing the peaks, the other just gently levelling the audio), but in a single compressor. It's also easier to control because it's less complicated, and it's a little more sophisiticated because the transition between low ratio and high ratio is smooth.

-Kim.

2009/08/03 - Don't fill the frequencies

Often I see advice given to fill up the frequency spectrum. That is, each instrument or part in the mix should fill a particular part of the frequency spectrum and that they shouldn't overlap.

The problem with this approach is that most sounds – especially natural sounds – have a relatively wide frequency spectrum. For example, a human voice might have most of its energy in the upper mids (perhaps around 2.5kHz), but will also have useful information ranging from below 200Hz to above 20kHz. Similarly, pianos, drums, guitars and other instruments have wide spectra. Filtering this information out to keep only the most important frequency ranges will result in a sound that is artificial and unnatural.

On the other hand, this might be a useful approach for sounds that have no relation to natural acoustic sounds – such as synthesisers. Audio generated by frequency modulation synthesis or granular synthesis is nothing like any acoustic sounds, and so there is no expectation of what it *should* sound like.

Regardless of the instrumentation of your mix, *it’s ok for sounds to overlap*. Consider a mix with voice and piano. Both instruments share the same frequency ranges, yet we can hear both instruments clearly. This is because there is much more to sound than frequency. There are characteristics specific to the instruments, such as harmonic structure and envelope, and there are characteristics specific to the composition, such as harmony and rhythm. It it these characteristics that allow us to distinguish between different instruments – not frequency range alone.

Problems exist when two instruments share the same characteristics. For example, if a song has both an acoustic piano and electric piano, it may be difficult to distinguish between them. In this case, the most natural way of addressing this problem would be to remove one of the instruments (or at least only have one playing at a time). Simply filtering the sounds won’t change the characteristics enough, and doesn’t actually change any of the characteristics that are causing the problem.

A similar problem occurs when there are several vocal parts in a song. In this case though, the effect of making the individual parts difficult to distinguish is *deliberate*. All the vocal parts are designed to blend together. If there is one part that needs to stand out from the others (such as the lead vocal), this part is mixed in the foreground, and the other parts and mixed in the background. Rather than separating them by frequency, it is much more natural to separate them by *depth*.

-Kim.

2009/08/05 - Backing Vocals

Recording backing vocals is a little different to recording the lead vocals. Rather than recording them forwards, then backwards, then forwards, I simply record them one section at a time – typically four or six takes for each part. I prefer a combination of syncronised harmony vocals (in time and harmony with the lead vocal) and unsynchronised ‘call and response’-type backing vocals (with different timing and rhythm to the lead vocal).

For bigger backing vocals, I’ll take the two best takes for each part, and pan them hardleft and hardright. The natural variance in intonation gives the part a very wide sound without being messy. It also sounds much more natural than using a single take and making widening it using artificial processes (such as delays or pitch shifting). When I want even more voices, I record different harmony parts and apply the same process. Sometimes I’ll go as many as three parts deep. This results in six total harmony tracks – three on each side.

The trick with harmony vocals is to go easy on intonation correction. Whether you use Autotune, Melodyne, GSnap, or something else, find a way to use it extremely subtly. The more in-tune the backing vocals are, the smaller the total effect is. Your job is to balance correctness with size. I find even correcting the vocals 50% has a significant effect – often too much! A lot of the time I’m happy to keep the backing vocals untuned, or tune one side and keep the other untuned. So long as the singer can sing reasonably well, it shouldn’t be too detrimental to the song. If the lead vocal is appropriately in tune, the backing vocals only need to add size and thickness.

On the other hand, if the lead vocal is weak (tuned or not), the backing vocals benefit from being much more in tune. In this situation, the backing vocals serve as a support for the vocal (and should be mixed appropriately).

-Kim.

2009/08/06 - Mixing with reverb 1

That's not to say you shouldn't spend time learning your tools. Quite the contrary – effective use of any tool require spending a lot of time with it, exploring how it works, learning how it can work for you. It's important to understand what the strengths and weaknesses of your tools are, and when to choose one tool over another. In audio engineering, this is the time you spend playing with your tool.

There are many, many different choices when it comes to reverb. There are different types – hall, room, plate, spring, reverse, gated, etc. There are algorithmic and convolution reverbs. Each developer and manufacturer has their own way of designing reverbs – a Lexicon sounds different to a TC. CSR sounds different to EOS. And even if you get this far, reverbs often have a lot of presets and parameters – a single hall reverb might have over a dozen different parameters, and over a dozen different sounds.

Given all this choice, it can be easy to lose sight of the end goal – to add ambience to a mix.

That's not to say you shouldn't spend time learning your tools. Quite the contrary – effective use of any tool require spending a lot of time with it, exploring how it works, learning how it can work for you. It's important to understand what the strengths and weaknesses of your tools are, and when to choose one tool over another. In audio engineering, this is the time you spend playing with your tool.

Back to reverb.

When it comes time to add reverb to your mix, stop and pause for a moment. Listen. I mean *really listen*. Listen to your dry mix, and *imagine* the ambience of the song (this is why it's important to get the balance and tone right before you start adding reverb!). Start with the basics – Does it need short reverb or long reverb? Should the mix be lush or dry? Then start to ask the more difficult questions – Should be ambience be deep or shallow? Should it be natural or unnatural?

You should know the answers to all these questions before you lay a finger on any reverb processors.

-Kim.

2009/08/06 - Mixing with reverb 2

Does it need short reverb or long reverb? Should the mix be lush or dry? Then start to ask the more difficult questions – Should be ambience be deep or shallow? Should it be natural or unnatural?

The answers to these questions should be based on the song.

Short / Long: This decision should primarily be based on the pace of the song. Obviously, slow songs will tend to need longer reverb and faster songs will tend to need shorter reverb. Keep in mind that pace is not the same as tempo. A song with a slow pace may still have quick tempo, and a song with a fast pace may have a slow tempo. The pace of a song depends on various factors, such as the density and syncopation of the rhythms (particularly drums and percussion) and the rate of change (both in the harmonic progression and overall structure).

Lush / Dry: This is more of a creative decision. It's a question of how much you want reverb to be a part of the sonic signature of the mix. A lush mix is usually more dreamy and evocative, whereas a dry mix has more clarity and immediacy.

Deep / Shallow: This is where things start to get difficult. The question of whether your mix should have deep or shallow ambience is a question of depth, and it's not directly related to the depth of the other mix elements. A mix with a big distance between foreground and background might still be best served with a shallow ambience. Similarly, a mix with a small distance between foreground and background might be best served with deep ambience. Deep ambience enhances the sense of depth and space in the mix, whereas shallow ambience enhances the softness and blurriness of the sounds. Mixes that are dry *and* shallow will typically have very little reverb at all.

Natural / Unnatural: Natural reverb best compliments acoustic instruments. It doesn't have to sound exactly like a specific room or acoutic space, but it should sound like an acoustic space that might reasonably exist. A natural reverb would also be appropriate when integrating sounds such as synths and samples into a more traditional instrumentation (such as vocal pop or a band). Unnatural reverb is best suited to mixes where most of the instruments have no acoustic basis (such as drum machines and synths), or where the sound of the mix is far from representing the acoustic sound of the instruments (such as modern pop rock). Unnatural reverb takes a studio production beyond a mere recording of an event to an artform in its own right.

-Kim.

2009/08/10 - Mixing with reverb 3

Does it need short reverb or long reverb? Should the mix be lush or dry? Then start to ask the more difficult questions – Should be ambience be deep or shallow? Should it be natural or unnatural?

Short / Long: Obviously, you'll need to adjust the reverb time. That's not all though – other parameters can be used to change the apparent length of the reverb. Adjusting the reverb size with the length can help keep the reverb sounding natural. Some algorithmic reverbs also have frequency multipliers can change the reverb time specifically for low frequencies. How you adjust these will depend on the mix.

A longer reverb time for low frequencies can be useful in warming up a thin mix. It can also make an orchestral concert hall sound more realistic. A shorter reverb time for low frequencies would be more useful in a mix that already has a busy low end.

Lush / Dry: A lush reverb treatment will require you to use *more* (wet) reverb than a dry treatment. High frequency cutoff and high frequency reverb time also affect apparent lushness. The more high frequncies in the reverb, the more noticeable it is. Some algorithmic reverbs can have their attack shape adjusted too – the parameter might be called something like "shape", "build", or "attack". Slowing the attack of the reverb can also affect its apparent lushness. A reverb with a fast attack is often more "invisible" because the bulk of the reverb energy is actually masked by the source sound – this can make the reverb seem drier than it actually is. Pre-delay can also serve a similar purpose in making the reverb more noticeable, albeit in a less refined manner.

Deep / Shallow: This is where things start to get difficult. Often it's not too difficult to decide if you want to mix ambience to be deep (far away) or shallow (close). It's more difficult to *hear* this depth, and most difficult to control it. Depth of ambience is the distance between the source sound and the reverberation. For deeper ambience, hall reverbs are best. To go deeper, use a larger reverb size. Lower diffusion settings make individuals echos more audible, which can help. For extreme depth, use some pre-delay. Make sure the reverb is appropriately quieter than the source sound (don't get too wet!). Conversely, rooms and plates are better for shallow ambience. Use a smaller reverb size and higher diffusion. High frequency response can also help – use darker reverb for deep ambience and brighter reverb for shallow ambience.

Natural / Unnatural: Natural reverb best compliments acoustic instruments. It doesn't have to sound exactly like a specific room or acoutic space, but it should sound like an acoustic space that might reasonably exist. Convolution is an ideal choice for creating the illusion of a specific acoustic space. Algorithmic reverbs, however, are better for creating a reverberation that is tailored to the mix. Hall or room reverb algorithms are best, and should be configured within sensible boundaries. Try some presets for ideas, and don't stray too far.

-Kim.

2009/08/11 - Effects on a send

Sends are an interesting component of mixer topologies. They allow a combination of mixing and parallel processing. When several channels have non-zero gain applied to a send, they are mixed together, sent through whatever processing is assigned to the send, and then returned on a new channel. The processing on the send 'hears' a mix of all the channels being sent to it. As the output of the processing is returned on a separate channel, it does not affect the original source channels. It also means this return channel can be managed separately to the other channels in the mixer.

The most common uses for sends is to add ambience to a mix using delay and reverb. This works particularly well for two reasons:

1. The ambience is added 'behind' the sound, so that the original sound doesn't need to be altered. This takes advantage of the parallel processing aspect of using sends.
2. Reverb and delay are usually gain-linear, meaning they do not change their sound with different input levels. Sending a quiet signal to a reverb will produce the same reverb sound as sending a loud signal to it (the only difference being the output level). Additionally, sending two different sounds to a reverb simultaneously has the same result as sending each sound on its own. This takes advantage of the mixing aspect of using sends.

Of course, reverb and delay aren't the only types of processing that can be used with sends. Modulation effects such as choruses, flangers or phasers are also common. They work because they also take advantage of the characteristics of sends – they work by *adding* a sound to the original sound, and they are *gain-linear* - they work the same way regardless of what the input level is.

Increasingly, it is becoming more common to hear of people using non-traditional types of processing with sends. Interesting things happen when using processing like compression and saturation on a send, because these processes are fundamentally different to *additive, gain-linear* processes like reverbs, delays or modulation.

The first thing that happens when using compression or saturation on a send is that the processed audio is mixed in with the unprocessed audio. In the case of compression, this will get you parallel compression – which usually requires two duplicate tracks or a specially-designed compressor with a wet/dry control. In the case of saturation, this adds some saturated sound to the original without significatly damaging the integrity of the audio.

The other thing that happens is that you have an opportunity to use the send as a kind of parallel bus. That is, you can send audio from several channels to a single compressor or saturator (which is then brought back into the mix in parallel with the original sounds). It's important to remember that it is a bus. For example, you might set up a compressor on a send, and send some kick and bass to it. Unlike a *gain-linear* process such as reverb, the compressor will respond *differently* to the kick and bass playing together than it would to the kick or the bass separately. The other thing to watch is that the compressor will respond differently depending on *how much* of the audio is sent to the compressor. Typically, the audio will be more compressed if more of it is sent to the compressor. Similarly, the audio will be more compressed when there are more active audio channels being sent to the compressor because the overall level sent to the compressor is higher. This can make mixing rather complex.

-Kim.

2009/08/12 - The case against compressed drums (articulation vs texture)

Don't overcompress those drums!

When drums are compressed, the body of the drums is brought up in level (relative to the transient). This creates the perception of longer sustain, making the drums sound bigger. By bringing up the level of the audio between the transients, there is more sound overall. This makes the drums sound fuller. Coupled with the right kind of envelope shaping from the compressor, this results in a drum sound that is interesting and exciting!

In the context of a mix, however, this can just as easily work against you. Within the context of a mix, drums traditionally serve as rhythmic articulation. That is, they are the 'spiky' hits that rise above the other instruments to establish the timing and the groove of the music. On a continuous scale between articulation and texture the drums are almost entirely articulation, whereas the other instruments are more texture.

When the drums are heavily compressed the longer, louder sustain between the drum hits adds more textural sound. This leaves less textural room for other instruments. This might be appropriate if the mix is quite sparse, but you want to make it sound thick and full. A heavily compressed drum group might only need a bass and melody to sound like a complete mix. On the other hand, it will be difficult to fit in a subtle layered pad or detailed background sounds. They'll have to be much louder to be audible, which muddies the mix and reduces its depth (because the background is not so far away).

A subtle and deep mix will be better served by shorter drums with less sustain. While these drums will sound weaker on their own, they'll be more appropriate in the mix. The space between the drum hits will provide ample space for bringing in other sounds, allowing either a subtle textural approach or a deep mix with a far background. The space between the drum hits is like a window through which the listener hears the rest of the mix.

-Kim.

2009/08/13 - Mixing with multiple reverbs

One way to contruct a subtle and complex ambience in a mix is to combine two different approaches to reverb. Going about this in an informed, deliberate way will result in a much more refined and appropriate sound than by simply stacking two different reverb algorithms (either in parallel or – heaven forbid – serial).

One way to approach it is to think about foreground and background. Often using a single reverb results in an ambience that sits primarily in the forground (resulting in a shallower mix) or in the background (resulting in a relatively dry foreground). Using two reverbs might allow a mix the benefit of both the foreground ambience (for softness and blurriness) and background ambience (for depth and spaciousness). One way to do this is to use a plate for the foreground ambience and a hall for the background ambience. This will be most coherent if foreground sounds are mainly (if not exclusively) sent to the plate, and background sounds are mainly (if not exclusively) sent to the hall. This approach is useful if the mix calls for a lush ambience with a three-dimensional quality to it.

Another approach is to combine short and long reverbs. This can be appropriate if the song calls for a long deep ambience, but there's no middle ground between too dry and too lush for some sounds. This way, some textural background sounds and feature sounds would use the long reverb and other sounds (particularly more percussive/articulative sounds) would use the short reverb. A hall or plate would be suitable for the long reverb, and a room or shorter plate might be suited to the short reverb. For a more unnatural sound, use a thick modulated hall for the long reverb and a non-linear reverb for the short reverb. This approach is useful for complex mixes that don't need to have a particularly realistic acoustic sound, such as electornic music and 'studio' music.

-Kim.

2009/08/14 - Alternatives to reverb

Reverb adds two properties to sounds – diffusion and depth. While there are many ways of changing the balance between diffusion and depth, there are times when a more extreme approach is required. Reverb may not be the best solution if a sound needs a lot of diffusion but very little depth, or a lot of depth but very little diffusion.

More diffusion, less depth

Diffusion is a way of blurring a sound, reducing its sharpness or distinction. A sound may need to be diffused if it needs to be pushed to the background or to fit it into a mix that is generally quite diffuse. This might need to be done in a way that doesn't add depth if the background of the mix requires a lot of clarity or if the mix is meant to be very shallow.

In these situations, processes such as chorus, microshifting, slap delay or even true doubletracking can be appropriate.

- Chorus diffuses the sound by adding a copy with constantly-changing pitch and timing. This can be appropriate if the sound will benefit from the added movement and the constantly-changing pitch is not distracting.
- For situations when the movement or pitch modulation are not appropriate, microshifting might be a better solution. This is commonly implemented as a pitch shift of a few cents down on one side of the stereo space and a pitch shift of a few cents up on the other side of the stereo space. This can give a very big sound that stretches across the stereo space, but doesn't have the modulated sound that chorus adds, and doesn't have the added depth or tail that reverb adds.
- Slap delay is shorthand for any quick delay with a delay time roughly between 30ms and 150ms. The delay time should be determined by the nature of the sound – the delay time and level should be set so that the delayed sound blends smoothly with the original sound. Slap delay can be useful when a sound needs less diffusion and more depth than chorus or microshifting, but not as much depth as a reverb might add.
- True doubletracking is a process of using two different takes of the same part being played simultaneously. The natural, human variations between the two takes will make them slightly different – different enough to create a different sound when both takes are combined. This is a popular technique for guitars and vocals because it can be used to create a very big sound while still sounding much more natural than applying chorus or microshifting.

Depth, no diffusion

Depth is a sense of distance – particularly a distance between the foreground and background of the mix. A shallow mix will have very little distance between the foreground and background, a deep mix will have a lot of distance between the foreground and background. Usually sounds are pushed to the background by adding both depth and diffusion, but in some cases it is useful to add depth without diffusion. A mix might need to be very deep, but also very sharp and clear (which would require diffusion to be minimised). In other cases,a mix might already be quite diffuse, and depth has to be created by using more obvious means (because regular reverb would be lost in the general diffusion of the mix).

In these situations, delay is often the most appropriate tool. Longer delays (>150ms) should work best. When tuning a delay for depth, rather than rhythmic complexity, it's often worthwhile tuning it by ear instead of snapping to the song's tempo. The sense of depth will come from hearing the echos between the notes. This may be difficult if a tempo delay is causing the echos to be perfectly timed to sound underneath foreground elements (so that the background echos are masked by the foreground elements). Making the delay more audible by tuning it in between tempo times will also allow the delay to be at a lower volume. This will enhance the sense of depth in the mix.

-Kim.

2009/08/17 - Masking

Masking is a little-understood concept that is important to composers and mix engineers. Essentially, masking is what happens when one sound makes it difficult to hear another sound. An obvious example of this is two instruments playing the same note, with one instrument sounding much louder than the other.

This can happen with notes or chords, where the voicing of one instrument covers up another, softer instrument. It can also happen with frequencies, where an element of one sound covers up an element of another sound. As with the example above, this happens when two instruments are playing the same note or frequency range and one is much louder than the other.

It can also happen when the notes or frequencies are not exactly the same, but nearby. The effect is particularly strong when both instruments are playing the same or similar parts, and the sounds blend very well. A common example is of distorted guitars and distorted bass. On its own, the distorted bass might have a heavy growl caused by a lot of energy in the lower mids and a crunchy fuzz on top. Once the guitars are brought in, however, the bass is reduced to a low-frequency rumble beneath the guitars. Even though the main energy of the guitars might be in the upper mids, it masks the upper harmonics in the distorted bass.

Another example is vocal harmonies. A song might have a section where the main melody is sung in parallel harmony – perhaps a third or fourth apart. If both voices are similar (sung by the same singer, in the same style, with similar processing), our ear will hear the upper harmony as being much more prominent than the lower harmony. The effect is sometimes quite striking – the lower harmony simply blends into the upper harmony.

These are both cases of the higher sound masking the lower sound.

Sometimes masking is useful, as it allows a sound to be thickened or deepened by adding other sounds to it. Other times it is undesirable as it makes it difficult for the listener to distinguish between the different sounds.

In the bass/guitar example, greater separation could be achieved by filtering or EQ so that each instrument contributes a unique sonic component to the mix. Alternatively, each instrument could be given a different depth. For example, the bass could be up front and the guitar further back in the mix.

In the vocal example, greater separation could be achieved by instructing the singer to perform each part differently – such as whispering one part, or perhaps singing one part forcefully. Better yet, have a different singer perform one of the parts.

-Kim.

2009/08/19 - What to do when you have too many plugins

The easy availability of plugins makes it easy and tempting to 'collect' plugins – resulting in a plugin folder with many different compressors, reverbs, EQs, delays and other effects (not to mention synths!). This can actually slow down the mixing procesess because of a perceived need to try out all the different options.

Rather than collecting plugins, a better approach might be to choose one plugin for each category or processor – one compressor, one reverb, one EQ, one delay etc. Be sure to choose a plugin that is reasonably multi-talented – a general all-rounder. Use it for a few projects. Get to know it – *really* get to know it. Through this process you'll get to know your own taste in processors. You'll get to know the full range of each processor, and you'll know where you're pushing the device to its limits. By this stage you'll understand your requirements.

At this point when you start looking for another plugin to compliment your existing set, you'll know precicely what you're looking for. If it's a compressor, you'll know if you need something smoother or something more agressive. If it's an EQ, you'll know if you need something more flexible or something with more vibe. Same for reverbs, delays and other effects.

-Kim

2009/08/20 - Tips for quiet recordings

When recording, it's important to control the sound that you're trying to capture, but it's also important to control the sounds that you're trying not to capture. Background noise can reduce your ability to process the sound appropriately and can frustrate your efforts to create a convincing mix. At worst it can ruin a recording.

It's important to deliberately consider noise, because it's easy to become accustomed to it and ignore it. Remember – even if you ignore it, the microphone will still capture it! There are two approaches to consider when attempting to reduce background noise:

Avoid the noise source

This is done by either removing the noise source, recording away from the noise source or recording when the noise source is quietest. Internal noise sources are often machines – computers, guitar amps, air conditioning systems etc. Turn them off as much as you can. Sometimes you can't turn them off – studio computers are an example of this. In these cases you should record as far away as possible – either place the computer in another room (with cables going through a hole in the wall, or under the door) or record in another room (with long cables for microphone and headphones). Sometimes the noise waxes and wanes – studios in busy neighbourhoods will be subject to traffic noise, for example. Learn the times that traffic is strongest, and schedule non-critical activites for those periods. Backups, organisation, cleaning and general maintenance can all be done during these periods. Schedule critical recordings for the quiet times – especially quiet or delicate sounds such as whispered vocals or finger-picked acoustic guitar.

Reduce the noise being captured

Once you've done as much as you can to avoid the noise sources, you should then focus your attention on reducing the extent to which the noise is captured on the recording. Start by identifying the direction in which the noise is coming from, and find the position in the room in which the noise is quietest. You might have to find a trade-off between a quiet position and a good room sound (sometimes the quietest position doesn't have a good room sound!). Then, if you can, put up barriers between the noise source and the microphone. This is easiest if the noise is actually in the room (such as a computer or climate control). Then consider mic positioning. Most microphones have a weak spot directly behind the microphone. Some microphones have different patterns that can be selected – these patterns change the sensitivity of the microphone at different angles. You'll get the greatest signal-to-noise ratio by positioning the microphone so the weakest spot of the microphone is pointing toward the noise source, and the strongest spot is pointing toward the sound you are trying to capture.

On the other hand, sometimes background noise is useful. It might be desirable to capture the background noise if it's part of the vibe of the recording. And outdoor vox pop and a live band both have (mostly uncontrolled) background noise that are part of the sonic identity of the sound.

-Kim.

2009/08/24 - Mastering versus mix-bus processing

It's a murky world, this mastering.

Mastering is a process by which a mixdown is prepared for distribution. Traditionally, this has been performed by a dedicated mastering engineer with specific skills and equipment. The esoteric skills and expensive equipment gave the mastering engineer a sort of mythical status. No-one outside the mastering studio really knew what the engineer was doing, other than 'making it sound better', or 'sprinkling magic dust on the record'.

Today, music technology is affordable enough that almost anyone can start creating recorded music with not much more than a computer. These recordings are often self-published online. In many cases, the audio that it heard by the listener is exactly what came from the original computer that recorded it.

Even though no dedicated mastering engineer is being used, there is still a 'mastering process'. Sometimes this mastering process consists of little more than rendering the mixdown and encoding to MP3. Sometimes it might be an elaborate process of mix bus plugins, comparisons with other songs, advanced multi-band processing and more.

The confusion lies with the deliniation of processes. Mastering is a process performed by people. It may also include modifying the audio, using means that are sometimes used in mixing. More confusingly, these means are sometimes used in mixing, but in a processing chain that is similar to mastering (such as using plugins on the mix bus).

So, to clarify:

Mixing is a process of combining the individual elements/instruments and balancing them so they all work together. The end result is a mixdown.

Mastering is a process of taking the mixdown and preparing it for distribution. The end result is music that translates well to all the expected playback scenarios.

The mix bus is a way to apply some mastering-type processing while mixing. Not all processing on the mix bus is actually mastering though – it depends on the *intent*. For example:

- Mixing into a bus compressor is *not* mastering when it's done to help the different sonic elements fit together.
- Using an EQ and limiter on the mix bus *is* mastering if it's used to balance the overall tone and 'loudness' so that the music sounds best in a mixed playlist.
- Using a limiter on the mix bus is *not* mastering if it's used to make the sound pump in time with the kick drum.

While mixing and mastering are two different processes, the use of the mix bus makes it possible to overlap them so that mixing and mastering are both done in the same environment.

Additionally, not all mastering processes can be applied on the mix bus. Trimming and fading is usually done in an audio editor. Encoding to mp3 or burning to CD are best handled by dedicated software and hardware. Preparing a collection of songs for an album usually can't be done in the same environment that was used to mix them.

-Kim.

2009/08/25 - Automation and expression

Many acoustic instruments have a wide variety of expressive possibilities. A performer usually has many ways in which the sound of the instrument can be subtly changed thoughout a performance.

For example, a guitarist often has more than one string available to play a particular note on. The strings can be excited with a pick or with fingers. The pick or fingers can excite the string at various points along the string. The other hand can 'stop' strings on the fretboard as they're being excited, or 'hammer' the fretboard to simultaneously stop and excite the strings. The fretting hand can bend (stretch) strings and slide between notes.

An instrumentalist can spend years learning how to control these performance techniques and then learning how to use them effectively. A seasoned performer will be constantly varying the way in which the instrument sounds in order to better support the music.

Electronic instruments are even more versatile in their sound. A single parameter can dramatically alter the sound into something completely different. Many synthesisers have real-time performance controls, assignable parameters, modulation and pitch wheels.

Use them!

When adding a new synthesiser part to a composition, think about how you can change the timbre of the sound throughout the length of the song. Try to do this in a way that supports the dynamic contour of the song. Make the tweaks in realtime if you can – use a MIDI controller with knobs anor sliders if you're using softsynths. If you're recording MIDI, you have the added flexibility of being able to record the expression in multiple passes – you don't have to do it all in one take like acoustic performers have to.

The same extends to effects processing. Subtly adjusting delay or reverb parameters over time can add drama and movement to song if it supports the overall contour. Alternatively, it can be used to reduce the dependency on more obvious dramatic movement in foreground instruments (the main instruments don't have to move as much because *everything* is moving). Automating insert effects such as saturation, modulation (chorus, flanger, phaser), or more esoteric effects (such as ring modulation or pitch shifting) can add movement to parts that are already recorded, or instruments that don't have a lot of expressive range.

-Kim.

2009/08/26 - The answer to "everything's been done"… But what is the question?

Don't try to be original. Try to be good. Originality will come without trying.

Often I see artists (not just musicians or composers) exclaim that they don't want to follow trends or 'sell out', and instead make 'original' art. They see the defining character of their work as being 'not something else'. They see being different for its own sake as a worthy goal.

I disagree.

Originality is a cop-out. It's what people announce their goals to be when they're afraid of unfavourable comparison. It's what people try to do when they don't have the confidence to take on something ambitious. It's their excuse when their work falls short of expectations.

Trying to be different is setting yourself up to fail. No matter what you try to create, your creations will be the product of everything you've experienced in your life. At best, you can come up with a *combination* of influences that hasn't yet been explored but your work will always be able to be deconstructed – revealing its influences.

Undoubtedly, these influences will be representative of your taste. They will will all have components that speak to you, that resonate, that describe your own personal understanding of music.

Using originality as a goal is not only a cop-out, but it is unnecessary.

Everything you create will be influenced by everything you've experienced in your life – an unique combination. Being original doesn't take a special effort – it's impossible to avoid!

Trying to be original results in flawed decision-making, where techniques or sounds are chosen merely on their being 'different' - *without regard of whether they support the creative direction or aesthetic of the song*.

Instead, try to make creative decisions based on what works best for the song. If the right choice is something different, this is good. If the right choice is something conventionaly, this is also good. It doesn't matter – so long as it's the right choice.

-Kim.

2009/08/30 - Emulations

I actually don't use the T-RackS compressors much. When I do, though, I choose them based on their sound. Personally, it doesn't matter to me what it was based on or inspired by or copied off… The only important thing is the sound. The 670 compressor could be called "Thick Gooey Compressor" and have an original interface, and it'd be the same to me.

Software is not in competition with hardware. There are many reasons for using hardware, and sound is one of them… but there are other reasons too – like workflow and studio strategy. I don't think anyone's going cancel an order on a hardware Fairchild because they can get a software version that sounds similar for $99.

It's great to have software that's modelled on classic hardware. The reason some hardware equipment is classic is that studio engineers have found them to be particularly useful. Using this gear as the basis for designing software gives us these same design principals that can make the software useful in a similar way. Still, I judge any piece of gear (software or hardware) on its own merits. It doesn't matter how well it 'models' another piece of gear – what's important is how it sounds on its own. I have some very accurate emulations that I almost never use, and some 'not-so-accurate' emulations that I rely on for every mix.

-Kim.

2009/09/02 - Lost in Tweakville

Do you ever get lost in Tweakville?

Do you ever sit yourself in the studio intending to make progress on a project, but instead spending hours playing and programming sounds or endlessly adjusting parameters? Perhaps you spent the last three hours auditioning hundreds of kick drums, before realising that the one you selected in the first three minutes was the best? Perhaps you wanted to add a synth part, but then got lost adjusting the hundreds of parameters? Maybe you wanted to add a compressor, but couldn't choose between the 30 different plugins you've got in your 'compressors' folder?

You've got too much gear.

There are too many options. Somewhere along the way you decided that the best way to improve you music was to acquire more gear… this is not always true!

There are two solutions: Use less gear and be more disciplined.

You'll be more productive (and have a better quality output) if you reduce your gear down to the minimum that you need. A thousand kick drums are useless if you only need to choose between the ten best. Dozens of plugins will hold you back as you spend hours browsing presets. Critically examine your hardware and software, your plugins, your sample libraries, your presets etc. Ask yourself if you really need that many.

The more options you have, the more time you spend evaluating those options, and the less time you spend actually making music.

Discpline is also important. Even with the bare essentials, it's easy to get lost in experiments. Keep yourself moving. If you can't find the right sound in ten minutes, chances are you need to rethink what you think you're looking for. If you find a sound that's 'almost there', chances are you won't find anything significantly better, no matter how much time you spend looking. If you get 90% of the way toward the sound you have in your head, you're doing pretty well. The remaining 10% can be achieved through processing and mixing. Keep moving. Don't get bogged down. Perfection is the enemy of completion. Accept that 'good enough' really is good enough. Stop playing and get on with making music.

-Kim.

2009/09/03 - Dynamic range and headroom

Noise floor

The noise floor of a system is the level at which the background noise occurs. In analogue systems, this will be the hiss and/or hum. In digital systems, this will be the point at which audio has less than one bit to represent it (audio at this level sounds like a crunchy mess).

Saturation point

The saturation point of a system is the level at which audio becomes noticeably clipped or distorted. In analogue systems, this is the point at which the system is overloaded and starts to behave non-linearly (often it's when the signal is distorted by 1%). In digital systems, it's 'Full Scale' – any louder and the audio is mercilessly clipped.

Nominal level

This is the measurement reference point. It's the level that we call 0dB (or sometimes, *unity*). In digital systems, this is always at the same level as the saturation point. In analogue systems, the nominal level is some distance below the saturation point: 18dB or 24dB for example.

The levels of the noise floor and saturation point are measured relative to the nominal level. For example, an analogue system might have its saturation point 18dB above the nominal level and its noise floor 72dB below ths nominal level. We say the saturation point is at '+18dB' and the noise floor is at '-72dB'. A 16 bit digital system has its noise floor 96dB below the saturation point, and because it's a digital system, the saturation point is also the nominal level: 0dBfs. Because the noise floor is 96dB below the nominal level, we say the noise floor is at '-96dB'. The level difference between the noise floor and the nominal level is also called the signal-to-noise ratio.

Dynamic range

The dynamic range of a system is the difference between the noise floor at the saturation point. In the above analogue example, the noise floor is at -72dB and the saturation point is at +18dB. Thus the dynamic range is 90dB. In the above digital example, the noise floor is at -96dB and the saturation point is at 0dB. Thus the dynamic range is 96dB.

The dynamic range of a piece of audio is the difference between the quietest level and the loudest level. If the dynamic range of the audio is greater than the dynamic range of the system, it should be compressed. This will reduce the dynamic range of the audio so that it can be adequately processed by the system.

Headroom

The headroom of a system is the level difference between the nominal level and the saturation point. In the above analogue system, the saturation point is at +18dB, thus it has a headroom of 18dB. In the above digital system, the saturation point is at 0dB, thus is has *no headroom* - in the traditional sense.

The headroom required by a piece of audio is the difference between the steady-state average level and the maximum peak level. In an analogue system the gain is often set so that the average level is at 0dB. In these cases, the headroom of the system should be greater than the headroom required by the audio – otherwise audible clipping will occur.

The situation is different with digital systems. Because digital systems have no headroom above 0dB, it is common practice to set audio gain as if the nominal level is actually much lower. Unfortunately, there is no standard practice or agreement for what the in-practice nominal should be. Bob Katz' K-System attempts to, among other things, set three standard nominal levels: -20dB, -14dB and -12dB. Each has a different trade-off- between available headroom and overall volume.

-Kim.

2009/09/07 - EQ – cutting vs boosting

Never boost, always cut. Or not…

EQ is one of the most important tools available to an engineer (second only the the volume fader). It can be quite a complicated tool to use, and it's not always easy to know how to apply it. Sometimes advice is given that 'cutting' (reducing the level of a frequency band) is inherently better than 'boosting' (increasing the level of a frequency band). Inevitably, the converse view is the cowboy "if it sounds good, it is good". Both camps rarely explain their reasoning, so how do you decide for yourself which approach to take?

The fundamental role of EQ is to change the tone of a sound. Whenever gain is applied to an EQ band, part of the sound is being changed. For the most part, this change only occurs at the area around the EQ band – the rest of the frequency spectrum remains unchanged.

- When reducing the gain of band ('cutting'), the changed part becomes quieter.
- When increasing the gain of a band ('boosting'), the changed part becomes louder.

Generally, reducing gain sounds cleaner because the changed part is de-emphasised in comparison to the unchanged parts. As a result, the natural character of the sound remains more intact, and the processing sounds more neutral and transparent. This is a useful approach to take when you want to maintain the general character of the sound, and the EQ is mainly being used to fit the sound in the mix. This works best when the original sound is well-recorded and already has an appropriate character for the mix.

On the other hand, increasing gain makes audio sound modified because the changed part is being emphasised over the unchanged parts. As a result, the changed part stands out and draws attention to itself. This approach works best when you want to change the character of a sound.

Of course there are exceptions to the rule. Often it's possible to boost the top end of a sound in a way that sounds natural – only brighter. Similarly, it's common to use dramatic low pass and high pass filters to make a sound radically different.

Personally, I almost always find that 'boosting' sounds wrong. The only exception is the top boost mentioned above – adding more high-frequency energy often doesn't affect the general character of a sound. Still, I probably apply 'cuts' about 80% of the time I use EQ.

Ultimately though, the question of how to apply EQ should be proceded by the question of what you want to achieve.

-Kim.

2009/09/14 - Reducing the distance between idea and output

The purpose of a studio is to create or record *music*. Hence, it should foster creativity. Certainly, a lot of creative work happens in the studio.

A lot of non-creative work also happens in the studio. Some of it happens 'out of session' – upgrading equipment, cleaning the ashtrays, backing up files, getting to know new gear, etc. Some of it also happens 'in session' – routing signals, setting up microphones, tuning up, auditioning sounds, etc.

Creativity is an enjoyable – and sometimes fleeting – state of working. In order to get the most of it, you should try to reduce the barriers to creativity. That means taking a good hard look at the non-creative work that happens 'in session', and moving as much as you can 'out of session'. Technology works best when it stays out of the way.

The options available to you depends on your studio setup and your style of working. Try to reflect on what sort of non-creative things you have to do in order to be creative. Here are some ideas to get you started:

Physical Arrangement

- Keep instruments such as guitars, keyboards, percussion, etc on hand. Within reach, if possible. You don't want to have an idea for a part and have to get a keyboard out of storage in order to realise it.
- Have front-ends ready for instruments – preamps, channel strips, even cables. Set up your studio so you can plug in a single cable to your instrument and be ready to go. Of course, you can adjust things if you want, but at least have *something* ready to capture that idea! Ideally, have instruments always plugged in and switched on – that way if you have an idea you can simply arm a track and start playing.

- Remove obstacles that you have to step over or walk around. You're less likely to grab that mic or patch that effects processor if you have to wade through piles of junk in order to get to it.
- Make sure your studio is a dedicated space. Having to share a space can make it difficult to get working when inspiration strikes.

Software Arrangement

- Set up a custom template for your DAW software, so when you start up a new project you already have your favourite synths loaded up (if you use soft synths) and you have channels already available for recording from external sources.
- If you use presets, make sure they're organised by sound category. This is especially important for samplers, which often have a tendency to group sounds by library instead of sound type. That means if you're looking for a piano sound, you might have to look in many different places to find the right one for the song. It's faster and easier to work if all your pianos are together, all your basses are together, all your synth leads are together, etc.
- In the same vein, make sure your plugins are organised by category as well. If you've got several compressors from different companies, you'll find it easier to work if they're all together.
- Reduce your choices. You'll work faster if you only have a few filter plugins instead of a few dozen. You'll choose a sound and get on with making music if you have one or two main synths and one main sampler instead of ten or twenty synths and half a dozen samplers with their own libraries.
- Keep your projects organised on your hard drive and move the finished projects to a separate folder. It's easier to find the project you need if you don't have to wade through a bunch of irrelevant files to get there. This is especially important if you've got multiple projects active at any one time.

-Kim.

2009/09/17 - Four basic principles

1) If you don't know why you need it, you don't need it.

This applies to almost everything in making music – Whether it be a lyric, purchasing an instrument or other gear, a particular approach to processing audio, or choosing songs for an album. It's shorthand for an approach towards a kind of minimalism – of not using anything more than necessary to achieve your goals.

Of course, this means you must know what your goals are. For example, you won't know if distorted guitars are right for your song if you don't have a clear idea of what your sonic palette is. You won't know if that extra bridge section of the song is helping if you don't know how long you want your song to be.

This isn't about a zen-style nothingness though, it's about knowing what you need. To do this you need to be mindful of the barriers in your workflow. You need to know what it holding you back. You also need to know what is available outside your project or outside your studio that may be helpful. This requires an attitude of constantly evaluating your own work (including finished products and workflow) and being aware of outside opportunities (whether they be artists, studios, gear shops, workshops, courses, etc).

2) If you can't hear it, you don't need it.

This is a specialisation of the first point above, but intended to apply specifically to hearing the differences between gear. Rather than asking other people whether Compressor A is better than Compressor B, try them for yourself. If you can't hear the difference, you should leave it at that. Trust your ears.

3) The volume fader is the most powerful tool available to you.

This should be your first port of call for finding the right balance between elements in a song. It sounds simple, but it's true. Before you reach for any EQ, compression, saturation or anything else, reach for the volume fader. Use it to position the sound where you want it in the mix. This should get you about 50% towards a final mix. *Then*, think about adjusting the tone with EQ. Getting the volume and tone right should get you about 80%. *Then*, think about adjusting the dynamics with compression. Volume, tone and dynamics are the three main aspects of a mix. Of these, volume is the most powerful.

4) It took you longer to ask the question than it would have to try it for yourself.

This is primarily in response to questions about specific usage of tools or techniques. It's actually quite quick to try something out for yourself. Plug it in, twiddle some controls, and pay attention to what you hear. If you don't know what to listen for, reconsider your reasons for trying it. See the first tip (above). If you try it out and still can't hear what you're doing, see the second tip (above).

-Kim.

2009/09/28 - Limitations vs creative direction

It can be difficult to find inspiration in the face of unlimited options. How do you pick a kick drum when you have a thousand to choose from (and are often told to layer a combination of kick drums)? How do you start writing a bassline when you can start in any key, and use any scale or mode? How do you choose what's right for your song when you can have every sound from acoustic guitars to full orchestras at the click of a mouse?

Modern technology is great because it gives us so many options and possibilities for creating music. Without a disciplined approach, however, it's easy to become paralysed by indecision – spending hours trying out sounds or experimenting with different grooves instead of making progress towards a completed piece of music.

An often-suggested solution to this paralysis is to impose limitations. This might be to write a song using a particular scale or mode, or to force yourself to only use a certain type of instrument, or to write a song within a certain time period. These are often good starting points to get quickly yourself out of a rut.

These types of limitations can, however, feel unsatisfying. This is because these approaches create a workflow that is defined by what it doesn't include – it still doesn't provide much in the way of direction. For example, giving yourself a limitation of only using a guitar as your sound source doesn't give you any direction of what to do with the guitar. It's still the same problem as before.

An alternative approach is to think about creative direction. Rather than defining a project by what CAN'T be done, instead define it by what WILL be done. Instead of providing boundaries, this approach provides focus. This works best if the creative direction you set is independent of tools or composition techniques. Don't think in terms of what gear you use or which notes you play. Instead, think about colour, texture, pace, etc. Even better, think one level higher – vibe, attitude, mood.

For example, you might begin a project where the focus is on creating "dirty electro rock". This doesn't limit the composition techniques or tools you use – you can use whatever you have at your disposal, so long as you use it in a way that evokes "dirty electro rock" for you. It could be anything from pure synths to mangled samples to live drums and guitars.

Another advantage of this approach is that a larger-scale project (such as an EP or album) will have a consistency and identity that makes sense to a regular listener. Non-geeks wouldn't know or care if an album was only made with softsynths or hardware. Ten songs with the same guitar could be as diverse as ten different genres. On the other hand, a collection of tracks all with the same higher-level focus will sound as though they belong together – even if the instruments change, the harmonies change, or the musicians change. This focus, combined with your own idiosyncrasies will form the sonic identity of the project. The stronger and better-defined the focus is, the more coherent the music will be.

You don't need limits. You need focus.

-Kim.

(Inspired by this post)

2009/09/30 - Four songs published

Two projects I've been working on have finally started to come to fruition.

Jarek

Windy Winds

Jarek is a rock band without a vocalist. With four guitars, a keyboardist and a drummer with only a kick and a tambourine, Jarek play a kind of atmospheric instrumental rock that shifts between evocative samples with ambient melodies and rocking out with heavy riffing. Their Myspace page is here.

The band came to me having recorded an album's worth of material at another studio. I mixed and mastered this single, and am negotiating to mix and master the rest of the album.

Erin Shay

1. Falling In Love

2. One More Last Kiss

3. Temporary Love

Erin Shay is a pop singerand songwriter. While her voice and melodies have a reassuring (and impressive) familiarity, she has an characteristic approach to composition – particularly harmony and tonality. Her Myspace page is here.

Erin came to me having written a collection of songs, but in need of a producer – as a third party to advise on songwriting and composition, and to guide the recording and mixing process. The drums were recorded at Debasement.

-Kim.

2009/10/05 - Gaining motivation

There are two seeds to successful and prolific work – **creativity** and **work ethic**. Work ethic dictates how you apply yourself to your work and consists of a number of factors, including **motivation**.

Motivation itself is 'desire that causes action'. It is not mere desire alone. It is not simply 'wanting' to do something. It is a kind of 'wanting' that causes action – a kind of 'wanting' that that makes you do things.

So what does it mean when someone wants to do something but isn't motivated?

To me, that implies there is a root desire, but somehow the action is not happening. To transform mere desire into motivation, there are two simultaneous strategies that must be taken -

1. Address the things that inhibit action. These are the things that get in the way. They can be psychological, such as a fear of failure or a tendency to be intimidated by large tasks. They might also be physical – such as not having the right equipment available on hand or not having sufficient time. Addressing these things might involve facing your fears (Just do it! Or start small!) or rearranging your lifestyle to accommodate.
2. Address the things that encourage action. These are the things that enable and improve the action. They might be psychological, such as the good feeling of having finished a project or the good feeling of positive feedback. They might also be physical – such as having a dedicated space and time for working. It's important to keep reminding yourself of the good feelings that you get from working. Likewise, it's important to keep the work space/time sacred and uncluttered from other distractions.

Some more studio-specific tips can be found here.

-Kim.

2009/10/07 - Reverb on the mix-bus

Under most normal circumstances, using reverb on the mix bus is no different to using a send on every track, with every send set to the same level. Usually this it not a good idea – it's better touse sends to apply reverb in different levels to different tracks. Some sounds can 'take' more reverb than others. Some sounds need more reverb then others to emphasise the depth in the mix. A send level of 0dB (unity – meaning the reverb is the same level as the dry sound) might still be not enough for sustained sounds like pads and organs. On the other hand, a send level of -21dB might sound extremely wet for staccato sounds or hand percussion.

Having said that, there is a place for mix-bus reverb. While it's not as refined or tailored as using individual sends, it is much faster. I've done it myself on occasion when I've had a project that's up against a hard deadline. Mix-bus reverb also sounds different to individual sends when it's placed *after* other mix-bus processing, such as compression or other dynamic effects (for example, NOT eq). Whether this sound is useful for you and worth the greatly-reduced flexibility is up to you.

Reverb in mastering is a slightly different matter. In this situation it's too late to adjust the reverb in the mix, so it can only be applied to the stereo mix. Reverb may also serve a slightly different purpose when used in mastering – to make all the songs in a release have a similar ambience. This might be particularly important on compilation albums or albums with a wide variety of sonic approaches.

-Kim.

2009/10/15 - [How to be more productive in the studio]

Get off the internet and make more music.

-Kim.

2009/10/15 - [How to reduce computer noise in the studio]

Get an acoustically-designed computer

An easy way to do this is to use a Mac. The latest Macs are already whisper-quiet. And you can also run Windows on them if you prefer to use Windows-only software.

If you don't want to use a Mac, another option is to use a purpose-built PC. There are companies that build these, but they tend to be quite expensive (and cost is usually one of the biggest reasons not to use a Mac).

If you'd rather build your computer yourself to keep costs down (or to get more value for your money), keep in mind that parts specifically designed for quiet operation can get quite expensive anyway. You'll be able to build a quiet computer, but it won't be for rock-bottom prices.

The bottom line is: expect to pay more for a quiet computer.

Isolate the computer

The next thing to do is to isolate the computer. How you do this will depend greatly on the physical layout of your studio. The best solution is to have the computer in a separate "machine room" (studios that record on tape often have the tape machine itself in a separate machine room). If you do this, make sure you get the highest-quality shielded extension cables you can find. Depending on your audio interface, you might be able to get by with only three cables:

- Firewire – dedicated to the audio interface
- USB – for mouse, keyboard, MIDI, storage devices, etc
- DVI – for your screen.

Failing that, try to place the computer in a separate enclosure. Here the trick is to balance quietness against airflow. Too little airflow may result in the computer malfunctioning from overheating – especially on hot summer days. Not a good look with clients! For home studios, you might try using a cupboard or cabinet. For professional studios, custom-made enclosures are ideal – especially if they include acoustic dampening, easy access to CD drives, managed airflow directed away from listening/recording areas, etc.

Obviously, the more you can start with a quiet computer, the less you need to physically isolate it. Likewise, the more isolation you can provide for the computer, the less you need it to be quiet.

Avoid recording it…

…by following these tips:

http://kimlajoie.wordpress.com/2009/08/20/tips-for-quiet-recordings/

-Kim.

2009/11/10 - ProRec Article – Reverb Types Explained

Consider this an extension to this article – I've just had another article published on ProRec explaining the different kinds of reverbs commonly used in recordings. There's good coverage of pretty much all common reverb types and good explanations, but the real jewels (in my view, at least) are the comprehensive audio examples with detailed explanations of what to listen for in each type of reverb.

Check it out here:

http://prorec.com/Articles/tabid/109/EntryId/345/Reverb-Types-Explained.aspx

-Kim.

2009/12/10 - How do individual tracks sound on their own before they're mixed?

When mixing, there are two different approaches to take when processing individual tracks (channels, instruments, sounds, etc) -

1. Try to make the track sound as 'good' as possible on its own, and then fit it into the mix; and
2. Pay no attention to the integrity of the track's individual sound and ruthlessly filter and EQ it so that only the most useful parts of the sound contribute to the mix.

Both approaches are valid. Which approach you take will depend on the type of mix you're working on and your own personal style.

Approaching each track individually can be a useful approach if you're aiming for a natural-sounding mix. This way, each instrument can be closer to its original (or expected) sound. The character of the sound it more complete and more nuanced. This approach would be achieved by using gentle shelving EQ instead of filters, wide and shallow parametric EQ, and a tendency to cut rather than boost. The downside to this approach is that it can make a mix sound very cluttered and crowded – especially if there are a lot of instrument or the instruments are badly-played.

Alternatively, you might approach each track as a collection of sonic characteristics to pick and choose from. This will lead to aggressive filtering and more surgical/targeted EQ, with a tendency to boost as well as cut. This way the mix can be very clean and controlled, and can be particularly useful in modern 'highly-produced' music such as electronic music. The downside is that the mix can become *too* produced, with each instrument sounding quite unlike its natural self. It can also result in mixes that sound thin, empty and gutless.

Personally, I take both approaches. Generally I try to keep foreground instruments as natural as possible. As I work my way further back to the background instruments, I get more and more surgical and ruthless. The more 'produced' a mix is, the more I'll tend toward aggressive filtering to cut out the unwanted characteristics of each sound and keep the desired parts. The more natural a mix is, the more I'll tend toward gentle shelving EQ to softly de-emphasise the unwanted parts of the sound and emphasise the desired characteristics.

-Kim.

2010/01/04 - Sweetening your mix bus, and why you shouldn't wait for mastering to do it

There's a case to be made for 'sweetening' your mix bus. Many mixes can benefit from some subtle processing to bring out the best qualities of the tone of the mix and to use dynamics to give the mix a more compact, controlled sound.

To bring out the best qualities of the tone of the mix, an EQ is the most appropriate tool. For this task, however, don't reach for your highly-flexible ten band fully parametric equaliser. Instead, go for something with character and vibe – not just in sound, but in workflow. The idea here is to use something with fewer controls, but where each control does something *interesting.* The recent Pultec-modelled EQ plugins are a good choice. The reason for this is that this tonal adjustment isn't a corrective task where surgical precision is requied. It's artistic, impressionistic. You're trying to be creative, to add colour, to make it interesting.

To use dynamics to give the mix a more compact, controlled sound, compression is the most appropriate tools. Unlike individual track compression, the best results here are achieved by being subtle. You don't want to completely change the dynamic behaviour of the mix. Instead, focus on less than 3dB gain reduction, and configure the compressor to simply ride the gain. Use high ratios when you want a pronounced effect, particularly on mixes with very little dynamics (such as rock music or dance music that is almost always at the same level). Use very low ratios for more dynamic music (coupled with a lower threshold to catch the lower-level audio). Faster attack and release times will produce a more pronounced effect, whereas slower times will be more more gentle and transparent.

The real tip here, however, is to do all this at the mix stage – not mastering. The mix is where you're focussing on creative sound adjustments, on making the song sound special. Mix bus processing clearly fits here. Mastering, by contrast, should be as transparent as possible – focussing on preserving the creative decisions that were made during mixing and translating that sound to the target playback format.

The best time during mixing to apply this sweetening is at the very last stage – after reverb and panning, just before rendering or recording the stereo mixdown. This is when you'll have the best perspective to apply processing to the overall sound. Otherwise you may end up chasing your tail in circles as further track-level changes necessitate mix bus changes.

-Kim.

2010/01/11 - Five compression mistakes and how to avoid them.

Compressors are complex tools and, like most other audio engineering tools, there are more ways to set them up 'wrong' than there are to set them up 'right'. If you're careful though, you won't fall into these common traps:

1. **Too much gain reduction**. You know you've done this when you've got tons on gain reduction and you're thinking to yourself: "It sounds great but I can't get rid of this massive click at the start of every transient." The click is from the attack time. Not only does it sound silly, but it will rob you of your headroom. Clicks like that are similar to deep bass – they're not very audible, but they can easily take up a lot of level. **Solution**: Either use less gain reduction (you probably don't need that much!) or use a limiter instead of a compressor. Another approach is to use a limiter after the compressor. Heavy-sounding compression is often the result of fast attack and release times rather than a deep threshold.
2. **Using compression to fix non-dynamic properties of sound**. You know you're doing this when the sound you're compressing has no dynamics to begin with (such as a synth bass/pad/lead). When you compare the sound with and without compression, the dynamics don't change, but the tone or harmonic content changes. In this case, the compressor is not the best tool for the job. **Solution**: Listen to the dry sound and consider whether you actually need a saturator or EQ. Next time, get out of the habit of inserting a compressor on every sound without first deciding if compression is what you really need.
3. **Using mix bus compression as an alternative to mastering**. You know you're doing this when you're rendering your mixdown to a file that will be burned straight to CD or encoded to MP3, and all you think you need to do is 'make it louder'. Mix bus compression has its uses, but it's not the right tool for achieving raw loudness. **Solution**: If you're in a rush and you don't care about quality, then use a digital limiter set to kill and call it a day. If you care about quality, either take the time to do it properly, or find someone to do it for you.
4. **Using mix bus compression as an alternative to working hard in mixing**. Don't be lazy! You know you're doing this when you're trying to use your mix bus compressor to change the sound of an individual element in your mix. Don't use mix bus compression to address a kick or snare that is too loud – it will have unintended effects on other mix elements too. **Solution**: Don't be lazy. Go back to those individual tracks that need fixing.
5. **Using side-chain compression to get two clashing parts to work together**. This is using a sledgehammer to crack a walnut. As above – don't be lazy! Side-chain compression can be useful as an effect, but it's certainly not necessary for simple mixing tasks like getting vocals and guitars to work together. **Solution**: Use tone and depth to separate sounds. More on how to do this later.

If you can steer clear of these common mistakes, you'll be well on your way to effective compressing!

-Kim.

2010/01/13 - [Suicide Songs (plus bonus album) now free]

Just a quick note to say that last year's album *Suicide Songs* is now available for free. I've also released a bonus album of new songs and additional versions.

A whole album as a bonus?

Suicide Songs was a project that took over three years – including writing, recording and post-production. I wrote 25 songs, recorded 15 of those, and chose ten of those to be 'remade'. These remakes were not mere remixes – they were complete re-imaginings of the original songs – often with very different tempo, instrumentation and vibe. Of the 25 recorded songs (15 originals + 10 remakes), 12 were chosen to form the final album. The remaining 13 lived in hiding for the last 12 months. Only now do they see the light of day.

Suicide Songs has been available for purchase since February 2009. Since then, I've moved on – choosing to work on new projects rather than promote it. As a result, sales have been low. This is a new year, and one of my tasks for the start of this year has been to 'retire' Suicide Songs – to finally put it to rest (so to speak). This project is the past, and it is dead to me. Rather than take it offline, I'm putting it up on my website for free.

Grab it here:

http://kimlajoie.com/Site/Suicide%20Songs.html

-Kim.

2010/01/18 - [How to use tone and depth to separate sounds]

A common problem with poor mixes is a lack of clarity between sounds. It's as if all the sounds in the mix are stepping on each other's toes trying to get to the front. As a result, the mix is messy and confusing.

This is often because the mix engineer could not decide which elements of the mix were more important than others, or which elements of each track were important to the mix.

Tone

I've written before about [filling frequencies]. Obviously that is a highly artificial way of separating instruments. Avoiding that technique, however, does not forbid you from deciding which parts of each sound are more important for the mix. Some sounds mainly contribute to the low end of the frequency spectrum. Some mainly contribute to the midrange, and others mainly contribute to the top end.

Listening to a sound in isolation (solo'd) can create an expectation that it should cover the entire frequency spectrum. This is fine if that is the only sound in the mix (or it's a very sparse acoustic mix), but in a denser mix it results in unnecessary crowding.

Instead, focus on bringing out the aspects of each sound that are most useful for the mix. If it's a growly bass, make sure you can hear the growl in the mix. If it's a bright synth lead or guitar, you probably don't need much low-mid energy. If it's small percussion, make sure it pokes through the top of the mix. The best (and most natural) way to focus the tone of each instrument is to use EQ to reduce the energy in the areas that are not so important. Sounds may become thin and/or caricatured, but often this is necessary to assemble a mix in which each sound plays its part.

Depth

Even after focussing on tone, you may find yourself with several instruments that occupy a similar tonal region. Classic examples are kick and bass, guitars and vocals, drums and percussion. In these cases where it is not appropriate to separate the sounds by tone, you must separate them another way. Depth (distance from the 'front') is an excellent way of doing this (it's a shame that many poor mixes have no sense of depth at all). Skilful and artful use of depth can make many midrange instruments work together beautifully.

Using depth well, however, requires the mix engineer (and often the producer) to make decisions about which instruments will be featured (placed in the foreground) and which will be relegated to the background. This is often difficult because the mix engineer and producer, having uninhibited access to the most intimate folds of a mix, can hear the beauty in every sound. It is certainly a tough decision to be able to polish a beautiful sound only to bury it deep within the mix where almost no-one will notice it.

Perhaps an admirable trait of a great artist is the courage to throw away good ideas.

-Kim.

2010/01/25 - Eight ways to write effective backing vocals

Backing vocals are easily overlooked in the production process. After all, the lead vocal was hard enough to record and mix, why would you want to record a bunch of more vocal parts? Backing vocals are not always the best choice for a song or a production, but often they can add substance and reinforcement to the song's message. They can also make a production sound more polished and professional (a single vocals line on its own can sometimes sound lonely or underproduced).

When you want to use backing vocals, there are actually more options than simply telling the singer to perform the lead melody with different notes…

In sync

This is the simplest way to write backing vocals. When the backing vocals are in sync with the lead vocal, the timing is the same and the effect is of harmony reinforcement. Use a backing vocal separated by a third or a sixth to bring out a colourful harmony (the bridge is often a good place for this). Use backing vocals separated by a fourth or fifth to add grounding and stability (the chorus is often a good place for this).

Out of sync

This is a bit more involved, and how you go about it depends very much on the nature of the song. When backing vocals are out of sync with the lead vocal, they 'break out' and are heard as a separate part with its own phrasing. There are many ways of approaching backing vocals like this. One of my favourites is to identify some key words in the lead vocal and stretch them out over several beats – either before or after the word appears in the lead vocal.

Non-word vocalisation

Another approach is to use non-word vocalisation (such as 'ah' or 'ooh') as part of the instrumentation. This can be very effective in bridging the textural/tonal divide between the lead vocal and the backing tracks (ever had a song sound like karaoke? This is the fix!). Long sustained notes can function like a pad – especially with several harmony parts layered. It's like a vocal pad or choir pad found on many workstation keyboards and synthesisers, but made from the voice of your singer! Short staccato notes can be effective in reinforcing a rhythmic aspect of the song. Be careful though – less experienced singers can have real difficulty with hitting the right intonation at the very start of each note.

Parallel motion

When writing a backing vocal to sit behind a lead vocal, the obvious way to contour the phrase is to follow the melody. When the lead vocal rises, the backing vocal rises. When the lead vocal falls, the backing vocal falls. This can be useful for reinforcing the shape of the melody, and is often useful in the chorus of a song.

Unlinked motion

Unlinked motion is a bit more interesting – this is where the backing vocal breaks away from the main melody and presents its own melody. This can be as simple as a slight modification of the main melody to add interest and melodic variety, or it can be as complex as a completely new melody (even with different lyrics and rhythms!).

Opposite Motion

An interesting hybrid of parallel motion and unlinked motion is opposite motion. This is where a backing vocal 'mirrors' the lead vocal. When the lead vocal rises, the backing vocal falls. When the lead vocal falls, the backing vocal rises. The effect can be ear-catching, but is difficult to pull off for long passages. It's not always easy to find suitable notes for the harmony that retain the mirrored shape of the melody and also hit notes that support the overall harmonic structure of the music. Additionally, this approach can sometimes be constrained by the range of your singer. Despite these difficulties, opposite motion can be effective in small sections – even single motifs.

Intermittent emphasis

Backing vocals don't have to be sounding for the same length time as the main vocal. In some situations, it's appropriate for the backing vocals to come in occasionally for certain words or phrases. This allows you to emphasise some parts of the main vocal over others. This approach is particularly effective for long verses or complex choruses, where it's easy for the listener to get lost. The backing vocals add some delineation and 'punctuation' to help make the song easier for the listener to understand. Of course, it's also useful for reinforcing particular words or phrases in the song that have emotional significance.

Call and response

This is a really good way of adding interest and energy to a vocal part with a lot of gaps in between phrases. The simplest way to do this is to have the backing vocals fill the gap with an echo of the main vocal part. Bonus points for using a different melody and different (but relevant) words. Using different words can also give you an opportunity to expand on the lyrical themes and add meaning. Don't fill in *all* the gaps, but do it in a way that supports the overall contour of the song (such as adding them to the second and last choruses).

Just two more quick ideas:

1. Don't forget to combine these different approaches. These are all techniques available to you. You should choose when it is appropriate to use them. Some songs won't require any backing vocals. Some songs are best served by only using one of the techniques above. Some songs will require a combination of these techniques in order to bring the best out of them. Always remember to support the lyrical content and the contour of the song.
2. These tips don't only work for voices – they also work for instrumental parts too! Even if you're composing music without words, you can probably find use for these techniques. For example, you might emphasise a techno lead synth with a second harmony part underneath it for the most intense section of the song. Or you might have a guitar solo being echoed and harmonised by a supporting keyboard part.

With these techniques in mind, try out some new ideas on your next song and see how they go. Some ideas might work, others might not. Either way – you'll learn something new!

-Kim.

2010/02/01 - How to convince yourself to invest in acoustic treatment

You need to acoustically treat your room.

You know it. You've read the articles, you've had people tell you. You already know that it's holding you back.

The problem is that you haven't done it yet. Despite you knowing how important it is, it hasn't happened yet. Maybe you're not sure how to do it, maybe that money has mysteriously disappeared into more plugins or instruments or other hardware. *Maybe it's just not sexy.*

If you're not quite sure how to do it, relax. It's not that hard. For a basic studio, you should start with some wall panels and some bass traps. The wall panels absorb and disperse the first reflections from your speakers. Imagine mirrors on your walls – anywhere you would see the reflection of your speakers when you sit at your mixing position is where you should put a wall panel. The bass traps hide in the corners and edges of the room. That's it. That approach will get you decent results for the first round of treatment, and will most likely be a noticeable improvement on your current environment (you can get more sophisticated if you want, but wait until you're designing your next studio for that).

If the money keeps mysteriously disappearing into more plugins or other gear, take a good hard long look at your setup. Chances are, you've already got plenty of gear. Chances are, you've got enough gear to last you the next few albums, at least. Don't kid yourself. How many more analogue-modelling synths do you need? How many more kick drum samples do you need?

Chances are, you need a new chair more than you need more music gear.

Despite what anyone else will tell you, acoustic treatment *is* sexy. It adds more sex appeal to your studio than any plugin or computer upgrade. Acoustic treatment impresses people who don't even know what it is, or why it's important (you'll recognise them as the ones who call it 'sound proofing'). Acoustic treatment is how people instantly know you're serious about your studio – especially if it's a modern computer-based studio which isn't necessarily brimming with hardware.

It's also how you know *you're* serious about your studio. Acoustically treating your room will motivate you and make you work more than you expect. It will make you excited to listen to music, it will make you excited to work on your own music. It will actually make you more productive.

And besides, there's nothing quite like telling people you spent $600 on foam!

-Kim.

2010/02/08 - Five EQ mistakes and how to avoid them

EQ is one of the most important tools available to audio engineers. Used correctly, instruments sound great and seem to fit together effortlessly. Used incorrectly, the mix resembles a battlefield with every instrument trying to destroy the others while fighting to be heard. One of the difficulties of EQ is that it is relative – the right settings depend entirely on the raw sound and the mix; there are no specific settings that will work *every time*. The key to using EQ effectively is to learn how to listen to the raw sound and identify what tonal changes will emphasise its best qualities and make it work in the mix.

1. **More of everything, everywhere.** There is tendency among inexperienced audio engineers to apply drastic EQ adjustments – particularly large boosts. Generally, the patter is like this: "I love this sound, but it needs more bass – let's boost the lows. That's great, but now it's not bright enough, let's boost the highs. Sounds great, but it's a little hollow, let's boost the mids." And so it goes. Obviously this does a lot of damage to the tone of the sound and it also wastes a lot of time too. **Solution:** Try to listen to the sound and decide what do you want to do to it *before* you apply any EQ. If adjusting the EQ in one tonal area is starting to reveal (or cause!) problems in other areas, think carefully before you start chasing your tail – sometimes you just need less EQ (and sometimes in a different place to what you initially thought). Always bypass your EQ from time to time and make sure you're actually improving the sound.
2. **Too much bass.** This is often a symptom of problems with your monitoring environment – usually either your speakers or your room. Or maybe there's not much wrong with your monitoring – you just happen to like the sound of too much bass. How you approach this situation depends on how much extra bass you tend to add. If it's gentle, and it helps you vibe with the music while you work, then you could consider *changing nothing*. **Solution**

1: Just make sure it's adjusted when your mixes are premastered or mastered… On the other hand, your mixdowns might have a wildly uneven bass response. You'll know this is you if the low end of your mixes sound strange and different when played on different systems. **Solution 2:** Bring some commercial reference tracks into your studio and spend some time comparing them with your mixes. If your monitoring environment is doing strange things, get more accurate speakers, invest in some bass traps. If you're mixing in a small room, look towards moving to a bigger room. If these solutions aren't practical, invest in a good set of headphones to use in conjunction with your regular speakers.

3. **Not compensating for your monitoring environment.** This is a more general case of the previous point. Be aware that you'll have a natural tendency to mix with an overall tonal shape that's influenced by your monitoring environment. For example, if your monitoring is weak on bass, you'll tend to mix bass-heavy. If your monitoring is strong in the mids, you'll tend to mix with the mids pulled back. If your monitoring has strong highs, you'll tend to mix dark. **Solution:** Be aware of this and make a deliberate effort to compensate for your natural tendency. Use more than one set of speakers or headphones for comparison. Listen to a lot of commercial music in your genre to learn how it sounds in your studio. **Bonus Solution:** If mixing with a *slight* tonal shift helps you feel the vibe (or in my case, I sometimes mix slightly dark because it helps me focus better and work longer) then keep doing it. Just make sure you remember to compensate in pre-mastering or mastering.
4. **Using specific frequencies or settings just because someone told you to.** Do I really nee to explain this one? Those frequency charts are just guides. That helpful advice is just one person's opinion, based on a monitoring environment you don't even know. Those EQ presets are just an example of what it can do. No-one else knows your mix like you do. No-one else knows the artist and the creative direction like you do. **Solution:** Use your ears. 'Nuff said.
5. **EQ even when you don't need it.** This is an issue that's not often spoken about. In all the flurry of trying to work out *how* to use EQ, it's easy to forget to yourself if you need to use EQ at all. Sometimes a part will sit perfectly in the mix without any tonal adjustment at all. **Solution:** Interrupt your "grab an EQ and start twiddling" workflow. Pause before you do it and ask yourself – 'Do I really need EQ here? What does this track *really* need?'.

Keep these tips in mind (or better yet – print them out and stick them to your studio wall!) and start to think about EQ more critically. EQ's a great tool for an audio engineer; don't be afraid to use it! Sometimes a track really does need some major surgery. Just be aware that it takes discipline and restraint to use it where you need it, as much as you need – no more, no less.

-Kim.

2010/02/15 - Your tools are not your competitive advantage

It's a war out there.

We're all trying to get ahead. There are too many of us looking for work and not enough clients or listeners to go around. We'll do anything to get our head above water – even if it means pushing down (or not holding a hand out to) our colleagues.

Or, alternatively, we're all on the same team.

There's a boundless and ever-growing number of clients and listeners who want to enjoy our music. The biggest challenge is getting the music in front of them. By working together we can reach more ears. By working together we can help each other improve our skills. We can encourage each other to work harder, producing work that keeps getting better and better.

Which world do you live in?

If you live in the first world, you probably keep secrets. You've worked hard to acquire your tools, develop your techniques. You might feel that it would be unfair to share your knowledge – for other people to be able to use your techniques or take advantage of your experience without having to earn it themselves. You might even feel that sharing your techniques would diminish your standing in the eyes (and ears) of your clients and listeners. Effectively, you want to hold yourself up by keeping others down. Maybe you even live in a slight fear that someone else will discover your secrets on their own – after all, that's how you discovered them.

Here are two uncomfortable truths:

1. Your clients and listeners don't choose you for your particular techniques. You might use something esoteric like multi-bus compression (multiple parallel compressors on sends), and it might even be a significant contributor to your sound. If someone else started using your technique, however, they wouldn't suddenly be able to steal your clients or draw listeners away from you. The same goes for tools.
2. Even if someone knew all your techniques and had all your tools, they wouldn't sound like you. They wouldn't somehow steal your musical soul, your originality or your creativity. They'd still sound like themselves and you'd still sound like yourself. If your techniques are useful to them, they'll use the techniques to get better at *their sound*.

Clients choose a producer or engineer based on many factors. The most important are not even remotely related to your tools or techniques – they are factors like your creativity, your availability, your ability to understand their creative vision, your ability to capture (and improve) the essence of their art, your ability to make them feel comfortable and creative. It's all creativity and relationships, with a good smattering of feel-good thrown in. Having a well-stocked studio merely helps them feel more secure in the decision they've already made.

None of these factors are diminished by the guy across the road having the same compressor as you, or the same preference for chaining three saturators in a row.

Of course, to be successful you must be good at what you do. This is not about techniques, though. It's about skills and creativity. This transcends techniques, and certainly transcends tools. It's about being able to understand creative direction and be able to choose the most appropriate techniques and tools to translate that direction into sound.

So if your tools or techniques are not your competitive advantage, what is? What is going to keep you ahead of the guy across the road?

- **A wide musical vocabulary.** This makes it more likely that you'll understand your client's influences and musical language. It also allows you to be more creative and choose from a wider variety of musical ideas. Of course, the greater your own musical language, the better you'll be able to come up with an idea that's perfect for the song.
- **An open mind.** This not only applies to music, but in dealing with people. Do your artists consider you to be someone set in your ways, or someone always willing to try something new? Are you artists comfortable suggesting strange and weird musical ideas, or do you have a habit of telling people they suck? If people know you to be flexible and fun to work with, they're much more likely to come back and work with you for their next project.
- **Work ethic.** Do you deliver? Do you come up with the goods by the deadline? Are you eager, focussed, hard working? A solid work ethic also rubs off on the people you work with. If you're dedicated and disciplined, your clients will treat you professionally. Your artists will take their own work more seriously (and they'll love you because working with you makes *them* more productive). You'll find your whole operation runs smoother and more efficiently. Best of all, less time and energy wasted on chasing people means more

time and energy for creativity.

- **The skills that pay the bills.** You have to consistently produce quality output. This is not about using any specific technique or tool, but being able to choose the right tools and techniques for each situation. It's about knowing your studio, knowing your tools and being able to get results quickly without wasting time. Your artists aren't going to be impressed if you spend half an hour tweaking a kick drum in front of them – you just killed their creativity. They came to you to capture their artistic vision and instead spent 30 minutes listening to 'thud thud thud thud'. Your clients aren't going to be impressed if they asked for an orchestral track and they receive something that sounds like a mid-90's GM synth. You need to know how to get the right sounds quickly.

So what of helping others? Will you still keep your secrets, or will you share them? Let me suggest that there's nothing to lose by sharing your knowledge, but there's certainly a lot to gain. Sharing your knowledge will help your friends, your peers, your teammates. By learning engineering techniques, they'll get better at realising their sonic 'vision'. The less mindspace they use trying to figure it out, the more mindspace they have available for being creative. And that results in better music for all of us.

-Kim.

2010/02/22 - Five secrets to making your mix louder

Don't dismiss this post yet! Even if you're in the 'more dynamics' brigade, these tips will give you clearer mixes that suffer less in mastering. That means better-preserved dynamics and higher fidelity!

For those of you who really do want your mixs SUPER LOUD, this tips will let you push more volume without your sound turning to mush.

1. **Go easy on the bass.** That includes sub bass, kick and melodic bass. It always tempting to turn them up, but the low frequencies really use up a lot of headroom (high peak level for the same perceived volume). The more headroom your audio needs, the less you can push it in mastering and the worse it will sound when it's pushed hard. First try to compare your bass levels with commercial reference songs. Listen carefully to the level of the low frequencies in comparison to the rest of the spectrum – you might find there's less than you initially thought! Also consider saturating the kick or the bass.
2. **Saturate those peaks.** Take a look at your mix bus peak meter to see if any tracks are 'poking out' of the mix – often it will be the kick or the bass. Used carefully, you can use saturation to reduce the peak level of your kick or snare tracks without reducing the perceived volume. Often peak level reductions of 6-9dB are easily attainable without adversely affecting the audio quality. Limiters are usually not so useful here because they'll tend to change the sound too much.
3. **Embrace the background.** Push some instruments further to the background. If you try to put too many sounds in the foreground you'll end up with an indistinct mush. This indistinct mush will quickly become even worse when you apply heavy limiting in mastering. Instead, try to identify the three or four most important elements of the mix (typically the snare, kick, bass and lead synth/vocal). Be bold and push everything else to the background! You'll get a mix that's more focussed and more powerful.
4. **Leave your stereo widener at home.** Stereo widening tricks might be fun to play with, but they'll rob your mix of punch and power. If you want those foreground sounds (snare, kick, bass, lead) to hit as hard as possible, stay clear of any stereo width manipulation. Some subtle widening is sometimes useful for special background effects, but remember –

if you do it, do it in moderation.

5. **Be careful of the lower mids.** The region between 100Hz and 1000Hz is the cause of many troubles. It's very easy to put a mix together that has a lot of mud build-up in that area. To get the ultimate clear mix, get brutal with an EQ! Make some big dips in the lower mids for all background instruments, and make sure you don't have any excess lower mids in your foreground instruments. You need to keep some lower mids, because that's where your body and thickness comes from. Here's a secret though – a mix with body and thickness only needs a few foreground instruments to have that body and thickness. To put it another way, a few fat foreground instruments makes for a fat mix. A lot of fat instruments makes for a flabby mix.

With these mixing tips you should be able to get a few more decibels of clarity in mastering!

-Kim.

2010/03/01 - The secret to full-sounding mixes

This applies to all the composer+producer+engineer types out there…

Have you ever felt like your mixes were empty? That they sound a bit incomplete? Perhaps you've compared your music to your favourite commercial references and realised that they somehow sounded thicker and fuller? You've got all the obvious parts in your mix – kick, bass, snare, lead, hats, pads, comp synths, etc – the same as your references, but somehow it sounds like you don't have enough.

What you're missing is the background.

The background is often made up of a lot of different sounds, each barely audible. On their own they sound quite small and insignificant, but together they form the sonic backdrop for your song.

You may not have paid much attention to the background parts because they're not sexy. The sounds are small, thin and don't draw attention to themselves. Listeners don't comment on them. It's understandable.

Sometimes it's hard to spend time working on the background parts. It's much more *fun* to focus on the big fat foreground – for the same reasons that your listeners focus on the foreground. It's naturally more interesting. Maybe you add some hats and percussion. Perhaps a synth pad or piano or something and call it a day. Besides, those drums need a bit more tweaking…

Here's a trick: Mute your foreground instruments.

That's right. Mute your kick, bass, snare, lead vocals, main melodic instruments… anything that's supposed to be the centre of your listener's attention. You're probably left with a somewhat unsatisfying background texture made up of only a few sounds (with a lot of dead space in between them!).

Now, go about adding some more parts! Fill in the gaps. Make it interesting. Build a musical texture that has character and individuality. Even if it's made up of loops, your choices and the combination of loops will make for a unique sound. If you want to add a bit more individuality, break out some crazy effects. Not just your regular EQs and compressors – this is the place for those strange modulation effects, sequenced effects, random beatslicers and other strange and wonderful contraptions. It's the perfect place to use that cool experimental effect that's too drastic for a main part, but you kept in your plugin folder 'just in case'.

This doesn't just apply to electronic music. If you're working on pop or rock, mute the drum kit, guitars, bass and vocals. There's lots of room for adding extra background parts. Perhaps add some more clean or distorted guitars. Try a different approach to micing your piano. Have some fun with rhythmic vocal growls. Create new percussion parts from kitchen utensils. This is your opportunity to try out a new instrument or recording technique. These are the kind of touches that give a song a unique character, a sense of individuality.

By adding in these background parts, your mix will become fuller and thicker. It will also have more character and texture. Even though your listeners don't usually comment on background sounds, they'll notice something different about your sound. Try it!

-Kim.

2010/03/08 - Making your song more dynamic

So, you've got a great groove going on. Now, all you need to do in order to expand those eight bars into a full song is add a layered intro, bring the vocals in and out, and add an ending. Right?

Not so fast.

In order to hold a listener's attention for several minutes, a song must do more than simply repeat the same ten seconds fifty times with different layers. Listeners are smart learners and they'll lose interest pretty quickly when they realise this is what you're doing.

So, you add yet more layers, or change the processing, or add a section without the kick drum, or something.

And it's still not working. You're looking for an engineering solution to a composition problem. You're trying to make your song more dynamic – this requires thinking about variety and drama. All this needs to be considered in context of the creative direction of the song. These are creative decisions – the engineering exists merely to serve the creative needs.

Variety

Variety works in conjunction with two other aspects of the song – length and coherence. For a given amount of variety, *a greater song length will result in more coherence*. Similarly, for a given song length, *more variety will result in less coherence*. Finally, coherence can be either good or bad. Let's unpack that.

A song without enough coherence will sound fragmented and unfocussed because there are too many unrelated musical ideas in it. A song with too much coherence will sound static and boring.

For a given amount of variety, a greater song length will result in more coherence. You hear this happening when you have a great 8-bar idea and try to expand this into a full song without adding new material. What sounded fresh and exciting as a 20-second snippet suddenly sounds boring and repetitive as a 4-minute song. Similarly, a collection of different ideas will sound jumbled and random when they're squeezed together in a sketch; give them space and time to breathe and develop, however, and you have the makings of an interesting piece of music that actually makes sense.

For a given song length, more variety will result in less coherence. You hear this when you have a 20-second loops repeated for a few minutes straight. This is a song with too much coherence – it's boring and repetitive. Adding more variety will reduce the coherence – making the song more interesting. Similarly, if a song sounds too compositionally unfocussed, it could be that there's *too much* variety. Reducing the variety (for example, by replacing some sections with variations or developments of other sections) will increase the coherence and make the song sound more focussed.

Drama

Being able to judge the right amount of variety in a song is an important skill to develop, but alone it's not enough. If you're trying to make your song more dynamic, it's important to consider adding more drama to the song. This involves two things – increasing the difference between the sections, and drawing attention to the changes between the sections.

Increasing the difference between the sections increases the scope and breadth of a song. Rather than simply adding or subtracting layers, think about giving some sections a very different texture, thickness, colour or pace. This also makes the song feel bigger because it covers a lot more 'territory'.

Drawing attention to the changes between the sections means paying particular attention to the transitions. For gradual transitions (such as builds to a climax), think about ways to increase the excitement and anticipation. A good way to approach this is to increase the *rate of change*. A section of music will be more exciting if a change happens sooner or faster than expected. Make more of the change occur in the second half of a transition. On the other hand, anticipation can be increased by making a change occur *later* than expected.

For sudden transitions, try to emphasise the transition… for example, for sudden transitions from a low-energy section to a high-energy section, try making the end of the low-energy transition even lower in energy, and the beginning of the high-energy section even higher in energy. Likewise, for sudden transitions from a high-energy section to a low-energy section, try making the end of the high-energy section even higher in energy, and the start of the low-energy section even lower in energy. Make the transition even more sudden.

So now you've got no excuse – make more dynamic music!

And if you want some advice on a song, email me.

-Kim.

2010/03/15 - Are you using ear-catching sounds effectively?

Ear-catching sounds are pretty easy to make. Grab a snippet of audio, throw some extreme processing on them, chop them up, and scatter the pieces around…

Interesting sounds are fun, but if you're not thinking too much about how they're used you're doing yourself a disservice. There's just as much impact to be created in placing the sounds as there is in having them at all.

Or to put it another way – It's not only what you've got, but what you do with it.

Many ear-catching sounds are quite short. Even if they're a few bars in length, they're still heard by the listener as an event, rather than as a whole section. This means they'll work best to emphasise the articulation in the structure of the song. In other words, use them to emphasise and draw attention to the changes and transitions in the song. For example, a stab or hit sound will have maximum impact at the start of a section – especially a section with a higher energy level than the section before it. Reversed sounds or sounds that build up will have maximum impact at the end of a section leading up to the next section.

Another way of using short interesting sounds is in place of a melodic motif. This works especially well when placed in between phrases in the lead part. For example, if the lead part is a vocal melody, you can place an ear-catching sound at the end of each phrase. In this way, there is a 'call and response' relationship between the vocal and the ear-catching sound. If you do it once, it draws attention to itself and signifies that there is something significant about that particular point in time (such as an important lyric or a transition to an important section). Alternatively, if you do it regularly it becomes a pattern and forms part of the musical 'language' and identity of the song.

Longer ear-catching sounds, such as sustained sounds, work well to differentiate a whole section – either adding energy to it or taking the listener to new sonic territory. If you're only using it as a layer in the section though, you're missing out on a lot of potential! Think about ways in which the sound can change over time throughout the section. The simplest approach is to fade it in or out. For example, you might have an interesting sound fade in throughout a verse. The verse starts as normal, but as it progresses the interesting sound becomes more prominent.

Don't just leave it there though… experiment with altering other characteristics of the sound instead of (or as well as) volume. Go beyond basic filtering too – think about changing effects parameters or synth parameters throughout the section. Make the sound have certain characteristics at the start of the section and different characteristics at the end of the section. Then think about making it change in a way that supports the direction and flow of the rest of the music.

Try these ideas out and keep them in mind and you should be able to give your music more interest and excitement!

-Kim.

2010/03/22 - When to separate sounds and when not to

When working on a song with multiple instruments, often you have to think about which parts will be played by each instrument. With larger productions, you'll tend to have more instruments than parts. In these cases, you'll need to think about when two different instrument sounds should be combined, and when they should be kept separate. When combining the sounds of two instruments, the effect is of a single sound that is a composite of both instruments.

How to separate

Separating instruments can be achieved in a variety of ways:

- Different register (pitch range)
- Different rhythms
- Different tonality (notes in the scale)
- Different volume
- Different sound character
- Different space (panning, depth)

For best effect, you should separate instruments using several of the above techniques.

How to combine

Quite simply, the techniques for combining instruments are the exact opposite to those for separating them:

- Similar register (pitch range)
- Similar rhythms
- Similar tonality (notes in the scale)
- Similar volume
- Similar sound character
- Similar space (panning, depth)

And again, for best effect, you should use several of the above techniques when combining instruments.

Of course, simply knowing *how* to separate or combine sounds is just part if the story – you must also know *when* to do it. When should you make two instruments more separate from each other? When should you try to combine them?

When to separate

You'll get the best results in separating instruments when the parts played by those instruments are already quite different. The best way to determine this is to think about the *function* each part has – what it's contributing to the song or the mix.

For example, you might have two melodic or harmonic parts in your song. You would be better off separating them if one is playing long slow notes and is only heard during the climaxes of the song, and the other part is playing small repeated arpeggios throughout most of the song. Even if both parts are played with similar instruments, in the same pitch range and with the same tonality, the two parts are definitely performing different functions in the song. To further separate them, consider changing the instrument or sonic character, the pitch range, or perhaps the depth of one or both parts. This will bring more clarity to the overall song. In other words – you hear them as different parts, so make them *more* different.

When to combine

The opposite is true when deciding when to combine instruments. If two parts are performing a similar function in the song, they're probably good candidates for being combined.

For example, you might have two parts that are playing short staccato rhythms (think of an arpeggio with gaps between the notes). The two parts might be played by different instruments, with different rhythms and different pan positions. Even still, they are both performing the same function in the song. To combine them, consider making the pitch ranges, pan position, or sonic character more similar. This will make your song more cohesive and and focussed. In other words – you hear them as similar parts, so make them more similar.

In applying these ideas, you'll bring more focus and clarity to your music.

-Kim.

2010/03/27 - Don't be alarmed, it's just a new theme…

The old one was getting a bit tired, so I've upgraded to The New Black.

-Kim.

2010/03/29 - Are your mixes too wide?

It's easy to go overboard these days.

Powerful computers and free plugins of every variety make for a very real embarrassment of riches. Nowadays most home studios are limited more by experience and skill than any lack of technical capabilities. And so, we have a tendency for people to overuse their tools. Many tools (especially compressors, exciters, saturators, etc) effectively have a single controls – 'more'. Turning it up gives you more of the effect. The *really* diabolical aspect is that adding a little bit often makes things sound 'better' (at first, anyway). So if a bit is good, then more must be better, right? Of course you know it isn't, but once you start *playing* with the controls, your mind recalibrates itself to what sounds normal. Even if you're deliberately trying to be moderate, you'll find it's all too easy to lose your bearings.

How many times have you wasted time trying to find the 'sweet spot', only to realise later that the unprocessed sound was better after all?

Stereo widening effects are a common example of this. A little bit often makes a track sound more impressive – wider and more expansive. After hearing the stereo widener, bypassing it can make your mix sound suddenly smaller and (more) lifeless. But how much is enough? How do you know when you've applied too much?

As usual, there's no single answer that's right for everyone. You need to consider where your music will be played. Different targets and/or media will require different approaches.

For example, if you're expecting your music to be played in uncontrolled stereo spaces (such as radio, shopping centres, or television) you'd best take a conservative approach to the stereo space. You don't know if people are going to be seated evenly between the two speakers, or if there even will be two speakers. I've had situations where my music was played back in mono – *by taking only one side* (not even by summing the two sides!).

On the other hand, if you intend your music to be enjoyed exclusively on headphones, or in cinemas, installations, or other environments with a controlled stereo space you can afford to use as little or as much stereo widening as you wish.

The difficulty comes when you're targeting a range of playback environments. You might have your music available for download – it might be the kind of music best enjoyed on headphones, but you'd also be happy for it to be played in cars, on computer speakers, or even iPod earbuds shared with a friend. In cases like this, you need to find a middle ground – where the amount of widening doesn't compromise the audio quality in adverse environments, but is still sufficient to express the creative intent of the song. You'll have to regularly check our mix in mono – not just both sides summed, but each side individually. Check your mix on headphones, on large speakers, small speakers and earbuds.

It's a compromise.

-Kim.

2010/04/05 - Five ways to make space in your mix

Running out of space in your mix? Want to add more parts without being buried in mud? Simply want a clearer, cleaner sound? Check out these techniques:

1. **Reduce the mids and low mids**. This area will add a lot of mud to your mix if you have a lot of instruments. It's not necessarily that all your instruments have energy focussed here (although they might!), but that having a lot going on in the mids and low mids gives a *feeling* of mud. Having strong mids or lower mids in just one or two instruments can produce a sound of warmth and body, but more than that is usually too much. If you want to create space in your mix, clear out the lower mids especially, leaving only the essentials.
2. **Don't squash the dynamics**. Dynamic space is very important. Natural dynamics and transients give instruments room to breathe. It also makes more space in the mix (for other instruments, or just for space' sake). Squashing the dynamics through overcompression, limiting or saturation makes individual sounds bigger, but sucks the life and air out. Of course, compression is often a useful effect, but be clear – the more compression you use, the less space you'll have in your mix.
3. **Push sounds further to the background**. I've written a lot about depth and effective use of background. With a deliberate approach to depth, you can draw focus to the most important elements of a song and still have a lot of space (or room for more instruments).
4. **Use panning effectively**. Personally, I've not a big fan of panning, but it's certainly a tool that, if used effectively, can enhance the space in a mix. Try mixing a song entirely in mono (or at least with every instrument panned centre), and then apply panning at the very last stages of the mix. You'll hear the space open up in front of you.
5. **Consider composition techniques**. Although this post is mainly focussed on engineering, composition has as much to do with creating space as mixing. Rhythm in particular can have a significant effect of the sense of space in a song. You won't have much space if everything is playing all the time (the effect is similar to the engineering approach of making everything louder than everything else). Instead consider restricting some instruments to off-beats, syncopated rhythms or using rhythmic counterpoint. Similarly, consider the pitch range of your instruments. Greater pitch range and mobility will open up space.

So next time your song is sounding too crowded, try this techniques and you'll be on your way to adding more space.

-Kim.

2010/04/12 - How to add more excitement and energy to your music

So, you've got your song planned out, the main parts are in place, it's humming along… but you're just not feeling the push. It's not making you sit on the edge of your seat – it's making you sit back (or worse, turn away…).

What you're missing is **excitement and energy**.

If your first move is to reach for a compressor, or harmonic exciter (surely an 'exciter' adds excitement?), or another serving of high-frequency boosts… pull your hand back. You're on the wrong track. Using these tools will make the song more exciting than it used to be, but if you apply them throughout the whole track, it won't make any difference to a listener who isn't familiar with the previous version.

Similarly, you might be tempted to add more 16th (semiquaver) hihats or other percussion. And similarly, it'll make the song more exciting than it used to be, but it won't make any difference to the listener.

Why is this?

It's because your song still has the same energy level from start to finish. Or, it has the same range in energy levels from start to finish. What happens is that in the first 20 seconds (or thereabouts) the listener becomes accustomed to the energy level in the song. It sets their expectation for the rest of the song. If the energy level of your song does not vary much, it will be lacking excitement – *even if it has a high energy level*.

What actually adds excitement? **Change**.

A dynamic song structure will add more excitement than any compressor or hihat rhythm. Think about the highs and lows, the ebb and flow, tension and release. I won't go into too much detail – I've already written plenty about structure here:

http://kimlajoie.wordpress.com/tag/structure/

Another way to add excitement is to use rhythm. A 'four-on-the-floor' rhythm commonly used in dance music is about as unexciting as it gets. Even if you have to use this for your kick drum (due to stylistic constraints), there's a lot you can do with the other instruments to add excitement. Add excitement by making musical events (notes, phrases, sounds etc) come in earlier than the listener expects. This can be in the form of accents that come 'before the beat', or repeating patterns that shift and change leading up to significant moments in the song.

Drum fills are a good example of this – they add excitement leading in to a significant moment. They work because the listener hears the drum elements (typically snare, kick and crash cymbal) earlier and more frequently than expected. They shake up the listener's expectation that the previous rhythm would continue. Experiment with taking a similar approach with other instruments too – shake up the established patterns at key points in the song. Making them faster and denser will help add excitement. Also, think about other techniques too – change the pitch, the timbre (brightness – open that lowpass filter!), the harmony (minor/major/etc), the interaction with other parts… there are endless possibilities.

With a bit of practice, you'll soon be adding so much excitement to your songs that your listeners will have trouble sleeping at night…

-Kim.

2010/04/19 - What's wrong with transient shapers?

Transient shapers are processors that adjust the dynamics of a sound. Rather than changing the dynamic range like a compressor, transient shapers operate only on the initial onset of the sound – the *transient*. The initial smack of a drum. The plink of a piano. The pick of a guitar or bass. They don't work with sounds that don't have a sudden start, such as vocals, violins, or synth pads. Transient shapers can either bring out the transient – making it louder, sharper and more prominent. They can also reduce the transient – making it softer and duller.

The tricky aspect to consider here is that the psychoacoustic (perceived) effects of a transient shaper can be similar to those of other tools.

For example, both EQ and compression can also be used to make a sound sharper or duller. Depending on the tone and envelope of the sound, an EQ or compressor can also be used to enhance or reduce the transients. They certainly can be used to make sounds more or less prominent.

In fact, for most day-to-day mixing tasks, channel EQ and compression offer almost all the sound shaping tools you need.

So why use a transient shaper?

If you *only* want to adjust the transient, EQ and compressors are blunt tools. Using a static EQ setting to boost the upper mids might bring out the ‘pluck’ of a guitar or ‘smack’ of a drum, but it will make the whole sound brighter. Similarly, using a compressor to adjust a transient will also affect the decay and/or sustain of the sound as well. Compressors are also level-dependent, meaning they process individual notes differently depending on how loud they are. This means that a dynamic performance will be treated unevenly – which is exactly what you want if you’re trying to control the dynamics, but not desirable if you’re trying to control the transients.

A transient shaper, by contrast, will process the sound while keeping its tone and dynamic behaviour intact. Most good transient shapers also operate independently of level, meaning they should apply the same amount of change to the transient, regardless of how loud or soft the sound is. Transient shapers are a subtle tool, and are best used when regular EQ and compression tools are unable to be subtle enough.

As always, a clear understanding of your tools will help you create the sound you’re imagining.

-Kim.

2010/04/26 - Four ways to use mid/side EQ

Several EQs now have a mid/side mode. This opens up a lot of possibilities, but can be difficult to use effectively. Instead of simply tweaking the sound or the range of the controls, mid/side mode completely changes how the EQ behaves and sets new rules for how it can be useful and effective.

It helps to stop thinking about mid/side EQ as an equaliser – but instead to think of it as a surgical frequency-focussed stereo width adjuster. It works best on complex stereo material, such as groups or the mix bus.

1. **Mono bass**. Not just bass, but lower mids too. It’s easy – use a highpass filter or low shelf (with negative gain) on the side channel. If you’ve mixed well, this won’t actually reduce the level or impact of your low frequencies (especially the ever-critical kick and bass). Instead, it will add focus and tightness in a way that doesn’t detract from the overall perceived stereo width of the mix. Experiment with the frequency – you’ll find you can probably go a lot higher than you might have expected. Unlike simply collapsing the kick and bass channels, using a mid/side EQ (particularly with a higher filter frequency) will also catch the lower mids in other instruments. And instead of making space in the mix by reducing their level, the mid/side EQ maintains their energy by simply collapsing them to mono.
2. **Top end dimension**. This is achieved by utilising a high-end boost on the side channel. Usually only a small amount is required – less than 6dB. Doing this to a mix can add dimension and air without the harshness of other tools (such as harmonic exciters or other saturation). It can also help open up a ‘small’ mix without losing the focus in the lows and mids. Some mixes will benefit from a more balanced approach – instead of adding 6dB to the top of the side channel, try adding only 3dB to the top of the side channel as well as reducing 3dB from the top of the mid channel. Not all mixes will benefit from this – it will sound more like a regular EQ boost if the top of the mix is already quite wide.
3. **Focussed vocals**. This can be done by reducing the width of the midrange. As with the above two tips, the most transparent way of doing this is by adjusting the side signal (by

applying a dip using a parametric band) while keeping the mid signal untouched. Doing this can reduce a lot of clutter surrounding the vocals, helping them to become clearer and more focussed. If you've got access to the mix, however, it's obviously better to do it the old-fashioned way. Consider using a mid/side EQ for this job as a 'magic trick' that you might resort to when your other options have run out.

4. **Giant lower mids**. This one's great for special effects – try boosting the lower mids in the side channel. It's an easy way to make something sound huge, without the associated headroom problems or (as much) mix mud. Of course, this technique is often as delicious as it is inappropriate, so have fun with it but remember to go easy in the final mix. A little bit goes a long way.

You'll notice that all these tips focus on making changes (either boosts or dips) in the side channel while leaving the mid channel (mostly) untouched. This is deliberate – it allows the width to be changed in a way that doesn't destroy the overall balance of the mix.

With these tips and a bit of practice, you'll be soon finding your own uses for mid/side EQ.

-Kim.

2010/05/03 - How to get out of a rut and rediscover inspiration

We've all been there. Halfway through a project, maybe even halfway through a song. It seemed like a good idea at the time, but right now it feels like the well has run dry. Nothing's grabbing your attention, nothing you try sounds good. You keep coming up with the same tired clichés and you're over it.

What do you do? Throw in the towel? Give up? Browse some forums *again*? Read up about defeating writer's block? Maybe. Here's what works for me:

1. **Don't take a break**. No, really – don't do it. You want to make music? How are you going to make music when you're on a break? Taking a break is shorthand for 'distract yourself'. It might feel good, but it's not going to get you back up to speed. Here's a secret – if you want to produce great work, *you have to work*.
2. **Get a good night's sleep**. I can't stress how important it is to be well rested. Being creative in the studio is a process of inventing new music. In order to do this effectively, your brain needs to be working at full capacity – able to recall, synthesise, transform and create new thoughts. Similarly, it's best not to be too stressed either – try to reduce other causes of stress in your life.
3. **Listen to different music**. A common cause for writer's block is that you've exhausted most of the musical ideas in your head. This is often a result of only listening to a limited variety of music. To refresh yourself with new musical ideas, seek out new music to listen to. It doesn't mean that you suddenly have to join a metal band if you were previously a lone dance music composer (or vice versa!), but bringing in even a few cross-genre influences can breathe a lot of life into a project. A large proportion of the most interesting music out there is influenced by a wide variety of sources.
4. **Force yourself to finish that song.** Yes. Force yourself. Bring it to the point where it can pass as a completed piece of work. It doesn't matter how – follow a formula if you have to. It doesn't matter if you're not inspired, or not 'feeling' it. It doesn't matter if it sucks. Just go through the motions. The important thing here is to establish a workflow, a pattern of finishing work. Don't let yourself leave unfinished ideas lingering. Real artists ship. You think creativity breeds productivity? Think again – productivity breed creativity. Once you establish a pattern of actually following through on songs and projects, it becomes easy to feed in your creative ideas. It becomes a conduit (not a barrier) for creativity.

5. **Take on a small project (and complete it)**. Sometimes writer's block can be caused by the intimidation of a large project. An album is a huge undertaking. No matter how much you want to make an album, no matter how many good ideas you have, it can be almost impossible to take the first step. It's daunting. Instead of procrastinating, take on a smaller project. Set yourself a 6-song EP, or even a 3-song demo. Set yourself a month, or a week, or a weekend, or a day. Set your goals, and then *follow through on them*. Get used to completing smaller projects before you take on big ones.
6. **Get intimate with a neglected piece of gear**. These are good times for musicians – instruments are more affordable and accessible than ever before. Chances are, you've got an instrument that you haven't explored as fully as you could – whether it be a physical instrument like a guitar or hardware synth, or a software instrument. Set yourself the task of getting to know that instrument through and through. Compose some music using only that instrument. Find creative solutions for sounds that the instrument isn't naturally good at. Guitars can make percussive sounds. Monosynths can be overdubbed and layered to form chords. Any sound can be completely transformed through effects processing.
7. **Buy new gear. Or don't, and say you did**. Yes, I said it. New gear is inspiring. The more different to your current gear, the better. A good gear acquisition will actually challenge you to think about making music in a different way. It's not just different limitations and different possibilities, but a new way of navigating the musical 'space' of rhythm and harmony. The more you challenge your existing ideas about how to make music, the easier it is to create new thoughts, new ideas and new music. Trying out new gear is also fun and motivating. Even if you can't afford new equipment, you can still try to *imagine* what your music might be like if you had an exciting new instrument. Try to push your current gear to the limits while you try to recreate the sounds you're imagining.

By following these steps you should be refreshed and invigorated, ready to produce your next hit.

-Kim.

2010/05/10 - What's holding you back?

What's holding you back?

What's your limiting factor? Your limiting factor is the single thing that's in your way – the thing that's stopping you from progressing and developing your skills.

Chances are, you know what this is. In most cases, it's not what's spoken about most (tools). It's a great time to be a musician or an engineer – tools are more affordable and accessible than ever. More likely, your limiting factor might be one (or some) of these:

- **Time** - This one's easy to identify. You probably find yourself muttering to yourself 'If only I had more time'. Or maybe 'If only less of this "life" stuff got in the way'. Or 'If only I didn't have to sleep'.
- **Discipline** - The is the alternative to time, for those people who have lots of time but still can't seem to get much work done (measured in *finished* projects!). You'll recognise this in yourself if you spend more time fooling about than you do actually making progress on musical projects.
- **Workflow / physical space** – Do you have to work in a cramped studio? Or maybe your workspace is actually shared (not a dedicated music-making space)? Bedroom studios are prime offenders here. Worse are situations where you have to actually unpack or set up gear at the start of each session and pack it up when you're done for the day.
- **Interesting musical influences** – You'll know this is a problem if you keep coming up with the same tired clichés in your music. Take a look at the last few albums you bought. If

they're all loosely in the same genre, you should think about branching out.

You get the picture. Tools are easy to get right (or at least adequate), but it's what a lot of people talk about the most. Try to move beyond the talk and take a good hard look at your work environment, your workflow and practices. Be honest with yourself. Chances are, buying another compressor or reverb isn't going to help you finish that project earlier or improve the quality of your output. If you're serious about your music, you might be better served by spending some time organising your studio space, or listening to fresh music, or simply spending more time working (and less time watching TV, or browsing forums, or whatever your particular timewaster is).

Another way of looking at it is this: If you have to ask someone else whether your tools are adequate, your tools are not your limiting factor. If you can't tell what's wrong with them, then they're not holding you back.

The better you are at identifying your limiting factor, the better you'll be able to push past it and make better music.

-Kim.

2010/05/17 - Why 'randomising' is not 'humanising'

How often do you see the terms 'humanising' and 'randomising' being used interchangeably? Or maybe you've seen someone ask how to make something sound more natural or human, and someone else suggests adding random variations to timing and/or velocity? Perhaps you've tried adding random variations yourself, only to end up with something that doesn't sound any more 'human' – just sloppy.

Of course, variations to timing and timbre are key to a natural human performance. The lack of variation is one of the very defining characteristics of samples and drum machines (when compared to human performance).

The important difference, however, is not merely that a human performance has variations – but that these variations have a pattern. They're not completely random.

If you analyse a human drum performance, for example, you'll find that some notes are always louder than others, some notes are always earlier than others. Other notes are always quieter, others are always later. This is *groove*. It's the complex combination of note emphasis, push and pull of timing, and rhythmic mode (which beats in the bar are usually sounding, and which beats are usually not sounding).

Of course, there will also be random variations as well – a human is not a machine! The variations will not only be in timing and velocity, however. There will be tempo variations. There may even be pattern variations (where some bars have a slightly different note pattern). These random variations, however, are not the defining characteristic of a human performance, and are often not the desirable characteristic we are looking for when we want to create the illusion of a human performance.

If not random, then what?

So, if we're going to add some deliberate variations to timing and velocity, what are we going to do? This is very much a choice that depends on personal taste and the needs of the song. Generally, I find that the following work for me:

- Stronger velocity on the beat (and weaker velocity between the beats). This works when a part needs to be stable and predictable, supporting the song.

- Stronger velocity on off-beats (between the beats) for foreground parts or transitions (such as drum fills).
- No timing offset on the first beat. This just makes it easier to think about where the bar starts.
- No timing offset on the beat (1, 2, 3, 4). This provides a solid, stable beat.
- Notes between the beats to be slightly late. You might already recognise this as swing or shuffle, but I'm talking about doing this much more subtly.

My usual groove within a beat is [0,10,20,10] – assuming 120 ppq[1]. This means that instead of every 16th note (semiquaver) being 30 ticks, the first two semiquavers are 40 ticks long, and the next two are 20 ticks long. This means that there's a bit of lag between the beats. The effect of this is that:

1. Strong notes on the beat have rock solid timing.
2. Strong notes on the beat have a bit of extra space after them, making them sound a bit bigger.
3. Weak notes lead up to a strong note on the beat sound like they 'speed up' leading up to the strong note, adding excitement and emphasis.
4. Strong notes *off the beat* have a bit more funk and groove.

Combined with lower velocities for the offset notes, this is usually not audible as an obvious swing or shuffle. For me, it adds just enough groove that I often don't feel the need to add additional variations – including random variations – to make a part sound human.

Humanising is more than just adding random variations to timing and velocity (volume and tone). The changes must be musical.

-Kim.

[1] PPQ: Pulses Per Quarter. One beat (crotchet) is 120 'ticks'. Half a beat (quaver) is 60 'ticks'. A bar (semibreve) is 480 'ticks'.

2010/05/24 - Frequency analysers and mastering

Sometimes frequency analysers can be used in mastering. Of course your ears should be the ultimate decision maker, but an analyser can be useful as a 'second opinion'. It can help sway you one way or another if you're unsure about something.

There are, however, some issues to keep in mind when using frequency analysers when mastering:

1. **There are differences between analysers**. Different analysers have different options and defaults for frequency tilt, time constants, resolution, etc. What one analyser shows as a straight line may look like a gradual rolloff on another. One analyser might show show short-term peaks differently to average level, whereas another might only show the peaks, and yet another might smooth everything out to show only averages. Configurations options might appear similar in some cases, but it can be difficult to know exactly how they're implemented 'behind the scenes'. **Solution**: If you're going to use a frequency analyser (or, really, any kind of analyser) make sure you pick one and don't use any others. Get to know how it responds to different signals. Don't get distracted by comparing its readings to the readings from other analysers.
2. **There are differences between songs**. Just because your reference song has a certain shape in a frequency analyser, it doesn't mean your song must have the same shape. Different voicings and dynamic behaviour will cause two songs to sound different with the 'same' frequency balance, and sound the same with a different frequency balance. Further,

non-technical aspects of music (such as structure, pace, harmony, etc) will also have an impact on how the audio sounds, which affects the frequency balance that is required in order for the song to sound balanced. **Solution**: Recognise that frequency analysers only measure some technical aspects of audio, and that music is *much* more than what can be revealed by a frequency analyser. The analyser is not a source of truth.

Ultimately, the only 'analyser' you should trust is your ears. Your tools can be helpful in some situations, but only if you understand their limitations.

-Kim.

2010/05/31 - Five ways to deal with an ugly vocal

Every once in a while as a producer or engineer, a project will come your way with one of *those* singers. With an… *unconventional* voice. Maybe they're inexperienced. Maybe their voice is just like that. Maybe they're doing it deliberately because they like it. Whatever the reason, you'll recognise this kind of project by that feeling you get when you hear the voice – *"What on earth am I going to do with this?"*

This is not to say that ugly vocals are *bad* – they're ugly in the sense of being unconventional, interesting and unique. The challenge is that it can sometimes be *very* difficult to make them work in a mix. And it's easy to get stuck or waste a lot of time with techniques that don't work. So next time you've got some ugly vocals to deal with, try think about these tips:

1. **Pitch correction**. No, don't turn your singer into a robot. It's worth trying, however, using stronger pitch correction than you normally would use. It won't make a bad singer any less bad, but it can help fit an instrument into the mix in a way that EQ and compression (obviously) can't.
2. **Low mids**. Pay attention to the lower mids – anywhere between 100Hz and 1kHz. Problems in this range can sometimes be quite difficult to identify. Sometimes all that's needed is a dip at 250Hz. Don't overlook (or overlisten?) the possibility that you might need *more* lower mids. This can be particularly true for thin or strident vocals. Sometimes a subtle bump in the lower mids can bring back some much-needed warmth or weight.
3. **2.5khz**. I almost always try a dip here. Be careful – this is where a lot of the voice's character is. Sometimes, however, there's a bit *too much* character in a singer's voice. Dipping around 2.5kHz can make a voice sound smoother. Too much, however, will make the voice disappear into mix – it'll blend *too* well and lose definition.
4. **More compression**. Another characteristic that a lot of ugly vocals have is dynamic peaks – the problem not being the tonal balance, but the strong peaks or wide dynamic range. In these cases it's worth trying stronger compression – lower threshold, higher ratio, faster response. It might make the compression more obvious, but it might not be a problem if the voice already has an unusual character.
5. **Learn to embrace it!** In trying to reign in an ugly vocal, don't lose sight (or sound) of the context. Try to capture, rather than suppress, the unique character of the voice. Don't get carried away in trying to *conform* the vocal – you'll end up destroying the sound, destroying the mix, and wasting your time. Instead, approach the character of the vocal as a critical contributor to the character and identity of the song, the album or the artist.

With these techniques up your sleeve, you should be able to do *something* with any singer that comes your way.

-Kim.

2010/06/07 - [Tuning the kick drum to the key of the song]

It's sometimes said that it's important to tune the kick drum to the key of the song.

While it's commonly said in relation to electronic music, it's certainly not restricted to that genre. Drummers tune their acoustic drum kits, no matter what genre they play. Often the individual kit pieces are tuned together so they sound consonant as a whole, similar to the way each string on a guitar is tuned in relation to all the others. Many other (membrane-impact) percussion instruments can be tuned as well.

When a drum kit is tuned, however, it is often tuned for tone, rather than any particular 'note' or 'key'. A drummer in a rock band doesn't retune the entire drum kit in between songs if there are two songs in different keys. Similarly, the drum kit doesn't change its tuning if there is a key change in the middle of a song, or if there are several different chords in the song.

Of course, there are different possibilities with electronic music. You could, if you wanted, change the tuning if the key of the song changes midway. You could even change the tuning of the kick drum every time there is a chord change.

But just because we *can*, doesn't mean we *should*…

As with an acoustic drum kit, the tuning of an electronic kick drum is more about tone than note. Certainly there is a consonance that occurs when the kick and bass are tuned similarly, but the tuning of the kick drum has a much bigger effect on its tone – its character and how it fits in the mix.

So, by all means consider the key of the song and the notes played in the bass when choosing and tuning your kick drum… but keep in mind that your choices shouldn't be solely based on this! In other words, don't let the kick or the mix suffer for the sake of tuning!

The other important thing to remember is not to hold yourself back from using a variety of chords or keys in your song. Being too tied to the tuning of the kick might cause you to avoid this – to the detriment of the song. It's more important to express yourself musically using all the compositional techniques at your disposal than it is to keep the kick in key with the bass *all the time*.

So next time you're thinking of tuning your kick drum, think about why you're doing it and make sure you don't get carried away!

-Kim.

2010/06/14 - [Do you have enough contrast?]

Contrast is essential in composition (and production and engineering too!). Any musical or sonic statement only makes sense in comparison with (or relief from) something different.

Of course, you know this.

Light and dark, smooth and harsh, quiet and loud, happy and angry, dense and sparse, consonant and dissonant, etc… You're nodding your head, but are you really implementing this?

Listen to (or think about) the song you're working on **RIGHT NOW**. Whatever's open in your sequencer (or at the top of the 'recent files' list). Whatever you're just about to spend your next session working on. Whatever you're writing in your sketchpad.

Is there enough contrast?

If there's scope for more contrast, don't be lazy – think up some ways to make the contrast more striking and more definite. Look at each section of the song and ask yourself – "What am I trying to achieve here? What is this section doing?”. *How can I give it more of what it wants to be?*

Be bold. Push yourself. You might even have to push past your fears.

Don’t settle for making the best music you can. Strive to do *even better.*

-Kim.

2010/06/21 - [If your song a jumble of noise?]

No, this post is not about mixing – it’s about composition.

Do you have too many unrelated musical ideas in your song?

While sometimes the problem is a lack of variety, other times the problem is *too much* variety. You’ll know you have too much variety when you have sections in your song that sound unrelated to each other – as if your song were made up of bits from other songs.

The problem with this is that it becomes confusing for the listener to understand the music. People are very good at finding patterns and relationships. Music lacking in patterns and relationships can easily end up sounding like a jumble of noise.

To make sure you’re not falling into this trap, think about the links and common elements tying together all the sections of your song. To improve coherence, consider aspects such as these:

- Instrumentation (sounds)
- Tonality (key, chords, scale)
- Groove (timing, rhythmic modes)
- Melodic ideas (recurring motifs and melodies)
- Contour (relationship in the overall structure in the song)

Of course, you need to strike a balance! Too much coherence will bore your listener, just as too little coherence will confuse your listener.

The next step is to be able to deliberately vary the coherence in a song. Not just randomly – but to deliberately choose different levels of coherence at different points in the song.

For example, you might choose to have a high level of coherence at the beginning of the song, in order to establish the musical language of the song and familiarise the listener with the principal musical ideas. Similarly, you might want to have a lower level of coherence in the middle of the song (or two thirds in – such as the bridge), in order to surprise and provoke the listener, and develop the song in a new direction. You might want to return to a higher level of coherence at the end of the song in order to create a satisfying conclusion and provide a sense of arrival or return for the listener.

-Kim.

2010/06/28 - Don't be lazy!

Are you taking shortcuts? Are you glossing over the details? Are you ignoring the intimate depths of your music, thinking no-one will notice?

Think again – they *will* notice.

Maybe not right away. Maybe they won't be able to articulate what they're hearing. Maybe even knowledgeable listeners won't quite be able to put their finger on it.

But they will notice. It makes a difference. Even if it's a small difference, it's this small difference that separates the great from the mediocre. It's the difference between *liking* to make music and *truly deeply caring* about the music you make.

Laziness in composition

Laziness in composition manifests itself in a lack of variation, a lack of contour, a lack of movement and detail.

Sure, it's ok to use loops. Loops are a great way of fleshing out the structure of a song – to get the length and pacing right once you've established the initial musical ideas, and to try out different arrangements or instrumentations.

But to leave the loops unmodified is laziness. You song wants to develop over time, to gain momentum, to move forward… so why are your loops static and unchanging? Even minor variations can make a difference. At the very least, include some variation leading into significant changes and transitions. Otherwise there's no difference between what you're doing and what the Garageband kids are doing.

It's tedious. Sometimes it's not much fun. But if you want to make the best music you can, don't be lazy. Roll up your sleeves, get your hands dirty, and get stuck in. Start by setting aside an hour or two just for getting into the nitty gritty. Do it.

Laziness in production

Laziness in production tends to manifest itself as a lack of creativity. You'll see this when a producer (such as yourself) makes the same choices for all the songs in an album – the same instrumentation, the same song structure, the same playing style, the same creative direction. Sure, there needs to be some significant unifying factor across the songs, but it needs to be balanced with a degree of exploration and creative freedom.

A lazy producer will also fail to push the artist and musicians to give their very best. An engaged producer will often record several takes, including individual sections and punch-ins, all the while providing specific guidance and coaching to the performer in between each take. A lazy producer will capture a couple of takes and say "it's fine". Mediocre.

If you are the producer in this situation, take a good hard look at yourself. And put some effort in. Do something interesting. Get engaged. You're there to provide creative direction, to make creative decisions. So make them. Make decisions, and make them creatively. And follow through on your decisions. Don't be satisfied with mediocre performances. Don't stop when you hit 'good enough'. Stop when you hit 'the best we can get'. Take no prisoners.

If you're the artist in this situation, pluck up the courage to tell your producer what your expectations are. Tell her/him that you're expecting to be pushed, challenged, guided. By someone competent. Otherwise, you may as well produce yourself.

Laziness in engineering

Yes, we've all been there. Let me count the ways.

- Using the same chain of processors, no matter what the sound source is.
- Presets.
- Making edits without crossfades or filling in gaps.
- Automatic pitch correction without note-by-note tweaking.
- Presets.
- Speaking about 'slapping' on a processor, or 'playing' with the controls.
- 'It's my favourite microphone, it sounds good on everything.'
- Not backing up the session files at the end of every session.
- Presets.

Stop cutting corners. Start caring.

Are you being lazy?

Ask yourself. Be honest. Think back to your last session – were you being lazy? Could you have pushed yourself, pushed the music to be even better? Did you settle for something less than what you could have achieved? Remember your thoughts and words when you were being lazy, and remember to catch yourself next time when you recognise those thoughts.

Now get back to work.

-Kim.

2010/07/01 - Mastering article on ProRec

It seems ProRec is back online, and they've published a new article I've written. It's a step-by-step explanation of the mastering process, using a specific song as an example. It's an account of 'a day in the life' - **Two And A Half Hours(including lunch) To Master A Single**.

- 10am-10:30am - Finding Reference Music
- 10:30am-11am – Technical Preparation
- 11am-11:30am – Technical Audio Analysis
- 11:30am-12pm – Fine-Tuning Level, Tone, Width and Headroom
- 12pm-12:20pm – Rest Ears, Have Lunch
- 12:20pm-12:30pm – Final Check, Trim and Fade, Export
- Bonus: Tips For A Better Mix Next Time

If you're interested in mastering your own music, check it out:

http://www.prorec.com/index.php/articles/mastering/item/56-mastering-workshop.html

-Kim.

2010/07/05 - Effects presets

Effects presets

Almost all effects processors come with presets. Most have more than a dozen, some even have hundreds. The number of presets is sometimes even used as a promotional line – as if it's a selling point. Surely if some presets are good, then more is better? Right?

Effects presets have their uses. They can be good for quickly exploring the range and scope of a processor, to hear what it's capable of doing. In some cases they're a good way for the manufacturer to demonstrate the processor's best features. In some cases they can speed up the production or engineer process if the engineer finds a preset that's close to what s/he is after, and adjusts it from there to reach the final settings.

So what's wrong with effects presets?

Effects processors, by their very nature, rely on the input signal for their sound. Their output depends entirely on the sound coming in. The 'effect' that an effect processor has on the sound is relative to the sound itself. It makes no sense for the settings to be determined by someone without hearing the input signal. To drive the point home:

1. It's like adding salt to your food without tasting it first. How do you know how much salt to add, or if it even needs salt?
2. "Bright Snare" preset? What snare? What style of playing? What song? What mix? Oh, it'll sound good with any snare? Really?

The exception here is for effects that change the sound so much that they effectively become just as important as the source sound. The obvious example of this is guitar amp simulators, but equally applies to a lot of creative distortion, filtering and delay effects. These types of effects are as much a part of the sound as the original sound source, and their presets function similarly to synthesiser presets – there is often as much artistry in how it's played as in how the sound is designed.

Ultimately, you have to do what's best for the music. Using presets will only get you partway there.

-Kim.

PS. As a side note, a *lot* of fun can be had by by incorporating presetless effects into your workflow! The best example of effects without presets is guitar pedals – particularly analogue pedals. Or things like this and this and these.

2010/07/08 - Obsession

This is a post about Obsession – not the personality trait, but the project!

http://kimlajoie.bandcamp.com/album/obsession

Over the last year and a half I ran a project called 'Obsession'. Its purpose was to showcase several of the artists I work with and explore the grey areas between genres. It features twelve artists – most of which are singers and songwriters. We spent the whole of 2009 writing and recording the album, and the first half of this year preparing for a live show (which was performed in May this year).

From a production perspective, working on Obsession was an interesting process. There was a wide variety of collaboration and working styles. Some artists wrote the whole song and then gave me free reign to do what I liked with it. Others came only with a few ideas and we developed those ideas into full songs together. Some artists were fairly 'hands-off', only coming to a few sessions to record parts and keep track of where the song was at, other artists came to every single session (either to contribute to the production process or to just watch and learn). Some artists were very clear about the creative direction they wanted, some gently guided the process, and others trusted my judgement and let me drive.

For me personally, it was a great change from working on my solo album. Where my solo album was 100% my own creative direction, Obsession was a deliberate effort to hand creative control to other artists, to make every song a collaboration. I'd worked with other artists on projects before, but not on something this big, and not on something where I was taking personal responsibility for the outcome.

From an engineering perspective, this was a particularly interesting project to work on because it was so diverse. There was everything from rock/metal to hiphop to folk to electronica – and there were some really inventive sounds too! It was a challenge to bring it all together and give the album a consistency despite all the different influences.

Listen to the album here: http://kimlajoie.bandcamp.com/album/obsession

Some interesting things to listen for:

What If – The giant 808-style kick combined with the light electronic percussion and NO SNARE (!) gives this track a really floaty feeling. Not to mention the synth pads and the great vocals by Jayne. When I was mixing this, I thought I'd drenched the vocal in far too much plate reverb, but listening to the song later I think I could have made it even more lush by adding delays too. Interesting how perceptions can change over time.

Addictive Personality – The extended guitar solo at 2:50 goes through a lot of processing and augmentation to really blur the line between live performance and synthesis.

All I Wanted – I love how the chord progression for the chorus changes slightly and develops each time we hear it. I especially like the double-time changes in the last chorus. The reversed guitars and subtle vocoder are great too.

By Calvin Klein – This was a really odd collaboration. The vocalist is Randall Stephens, who is primarily a poet and spoken word performer – not a rapper. It was actually quite an involved process to fit his words into musically-useful patterns!

Here I Go Again – The acoustic guitar solo at 3:01 was me playing my old beat-up acoustic guitar with dead strings. It was a real struggle to fit it in the mix by keeping the dynamics and noise under control. The 'pre-echo' background vocals in the last chorus work really well too!

When Night Passes – I bet you can't guess what the solo instrument at 1:57 is! If you listen carefully in the quiet sections of the song, you can hear the sound of people talking. It's a recording of people in a shopping centre that I made years ago, and adds a really interesting ambient texture to the song. That sound at the end of the song is reverb feeding back on itself…

The Cloak Of Night – We were planning to add a cello part to this song. It was a great idea at the time (and listening to the song now, it's still a great idea!), but the cellist we used wasn't gelling with the song. Just because you record it, doesn't mean you have to use it!

Stay With Me – I love the dirt in this song! The noise from the guitar, the saturation on the vocals and piano… It's certainly not a conventional pop sound, but it works really well to give the song a real sense of character. The flutes were a last-minute addition at the request of the artist, but ended up fitting in quite well.

Kissless – The drum loop is sent through two spring reverbs at 0:39. On one side is a a physical analogue spring reverb, the other is a digital model. I can't remember which side is which. There's also a sequenced Juno106 bassline throughout the whole song, though it's mixed so low I think I'm the only one who can hear it.

What Have I Become – Katie is one of the most dynamic singers I've worked with. She'll often go from a whisper from a scream in the one performance. This is often a real challenge for compression – to keep the whole vocal sitting well in the mix without undercompressing the quiet parts or overcompressing the loud parts, while keeping the whole thing sounding natural.

Beautiful Deception – So many words! Maxine kept running out of breath during the vocal recording session, and we had to make some edits to the lyrics just to get enough air in! There are so many lyrics that there wasn't time for a guitar solo or other instrumental break.

Perfection – The vocal effect at 2:43 is really interesting. It's based around an envelope-follower… but where envelope followers are commonly used to drive compressors and filters, this one was used to drive a pitch shifter (mod delay actually)! The main vocal is compressed (so it's all the same level), but the effect was triggered by the uncompressed vocal. You can ear the syllables that were sung loudest were pitch shifted the most.

I Love You Baby – I don't know where to start with this one. There are so many layers, so many cool tricks. I just went crazy here – lots of pitch shifted vocals, lots of live-performed distortion and filtering. I even indulged in some trance stutter synths! The Guitar solo at 4:20 is one of my favourite sounds. I've used it on a few projects (hear it on Star at 1:37 for example). Does anyone else use this sound?

World Order – I almost didn't include this on the album, thinking it was too avant-guard. Turns out a lot of people liked it, and it worked really well live! Loving the feedback – it almost sounds like a didgeridoo (especially at the end).

http://kimlajoie.bandcamp.com/album/obsession

-Kim.

2010/07/12 - Development and momentum

Development and momentum are two concepts in composition and production. They make longer term structure effective. They are the difference between a collection of sections in a logical order and a complete unified song that tells a coherent story.

Development

A song having a sense of development means that the listener hears the song grow and unfold as it progresses. This makes for a more compelling and engaging experience for the listener because there is a level of intrigue and surprise, simultaneously with a feeling of being taken along for a ride. When there is new musical material, it is not like changing the channel – it builds on previous material, appearing as the next extension. In some cases, this can come across as the original material growing out and becoming larger or more complex than before. In other cases, it can come across as additional detail being revealed – as if the listener is 'zooming in' and seeing more.

To give your music a sense of development, you need to think beyond musical structure being a collection of sections in a logical order. You need to think about each musical element. Not necessarily instruments or tracks – but *musical* elements. This includes:

- Characteristic sounds
- Melodies
- Rhythms
- Chord progressions

Think about ways in which they can be extended or expanded, and see how those extensions work as developments of the original material.

Another approach is to take a musical element that's already quite complex, and *reduce* its complexity. The reduced version becomes the 'original' – the form in which the listener first hears it. As the song progresses, bring the complexity back in.

Momentum

Momentum is a sense of moving forward. Think of it as using development with a deliberate *rate of chance*. The rate of change is key here.

Beginning composers often make music where the rate of change is too slow. This can be the case if each section is too long – even if the song has a good contour, and even if there's a good sense of development. When the sections are too long, the listener gets bored and stops anticipating the next section. In other words, you lose momentum. This happens regardless of how 'exciting' rhythms or loops are. Even if it's 150bpm high-energy techno – a minute of the same bar over and over again has no momentum.

At the other extreme, a rate of change that's too fast will confuse and disorient the listener. Instead of excitement, you'll end up with randomness. If the listener cannot understand the music, there's no anticipation and no momentum.

What rate of change is right? This is a matter of judgement, and different sections of a song will require different rates of change – depending on the contour of the song. As a composer (or producer), you have to develop your own sense of pace.

With a bit more work in giving your music a greater sense of development and momentum, you'll make your music more compelling and keep your listeners coming back.

-Kim.

2010/07/19 - Contour

Contour is the overall 'shape' of a song. While structure refers to the order and length of sections within the song, contour refers to how those sections relate, how they react to each other, and how they flow.

Contour includes the rises and falls in energy level, the establishment and return to main themes, and the development of musical elements. When a song has a well-defined and sensible contour, the listener will better understand the music and feel the anticipation and excitement as intended. When a song has a poor contour, the listener will feel lost and alienated.

Energy level

A song with a good contour will have variations in energy level. Periods of high energy energise and excite the listener, whereas periods of low energy provide relief and anticipation for the listener. Effective placement of high energy sections and low energy sections is an important consideration when designing the structure of a song. If the changes are too slow, you lose momentum and the listener's attention. The changes are too fast, you don't give the listener enough to recognise and latch on to.

Main themes

Musical themes give your listeners a way to remember parts of the song – not just after listening to the song, but *during* it. By establishing one or two main themes at the beginning of the song, you can then guide the listener through familiar material and unfamiliar material. A good song needs both, for similar reasons as needing different energy levels. Familiar material provides reassurance and recognition for the listener, whereas unfamiliar material provides excitement and development. Of course, too much of either will make for a weak song (too much familiarity becomes boring, too much unfamiliarity sounds like randomness). A good contour will require effective placement of familiar and unfamiliar material to guide the listener through the song.

Traditionally, musical themes are entirely melodic (or harmonic) – recognisable motifs, melodies, chord progressions, or other such material. Depending on your own approach to music, however, thematic material may also include characteristic sounds, or even distinctive effects processing.

Development

A song with good contour will unfold and grow over time and take the listener along with it. Not only that, but the development of the song will occur in a deliberate way throughout the song –

working together with the flow of energy and the placement of musical themes. I've written more about development here:

http://kimlajoie.wordpress.com/2010/07/12/development-and-momentum/

With a good understanding of contour, you'll be able to make your music more engaging and enjoyable for your listeners. More than simply being a collection of musical ideas, good contour will give your song shape and cohesion.

-Kim.

2010/07/26 - Bouncing to audio

'Bouncing' to audio is a process of rendering realtime generated audio to audio files. Typically, 'realtime generated audio' is software synthesisers, samplers, hardware sound generators, or even audio files being processed by plugins or hardware effects processors. After bouncing, these audio sources are turned into audio files on your hard drive. The audio files are a snapshot of how those sources sound – the same way a tape recording is a snapshot of a performance.

There are a number of different terms for this. Often you'll see it referred to as 'rendering' or 'exporting', or even 'loopback recording'. The term 'bouncing' harks back to multitrack tape recording systems, when the process involved re-recording audio from some tape tracks onto one or more other tracks. The audio was 'bounced' from track to track on a tape system.

Doing this can be a good idea for a number of reasons.

- **It can help conserve resources**. In a DAW environment, it can allow you to conserve CPU (by rendering a track that uses CPU-hungry plugins, then deactivating those plugins). In a hardware environment, it can allow you to use a specific piece of equipment – either an instrument or an effects processor – on many tracks at once.
- **It can make a project more portable**. By rendering tracks, you can bring the project files to another studio – even if that studio doesn't have the same plugins or hardware that you do. It can even allow a project to be shared between different DAW platforms or studios based on harware/software (and mixtures of both).
- **It can help you make decisions**. Rending tracks locks you in to a particular sound and performance. While realtime generated audio allows you to continually adjust the track (and for MIDI – the performance), rendering those tracks to audio files creates a snapshot that cannot be changed much (or without difficulty). This can be made part of a project workflow to mark the end of one stage and the beginning of the next stage.

Obviously, there are a couple of downsides. One is the track space. In a DAW environment, rendered audio files take up additional hard drive space. This is usually not an issue, because hard drives are cheap and high-capacity. It's more of an issue with hardware recording systems, because some have some very strict restrictions on how many simultaneous tracks are available at once.

The other downside is that it prevents further editing of the track – both the effects processing settings and (for MIDI) the performance. This is usually mitigated by keeping a deactivated copy of the original realtime generated track.

Personally, I use track rendering at two points in my workflow:

1. **When the artist brings their demo to my studio**. My artists work on a variety of platforms, so I ask them to render each track to bring them into my studio for further work.
2. **When using hardware instruments, hardware effects processors, or CPU-heavy plugins**. Obviously, this is to allow these tools to be used many times in a project. It also

allows projects to be recalled at later sessions (I use some hardware devices that are very complex and have no presets). I also use a CPU-heavy amp simulator, which I routinely render to audio as it's being recorded – because I prefer not to have restrictions on how many guitar parts I use (and it's no different to recording an audio file of a physical amp).

The decisions and if and when you render tracks to audio depends on your project workflow, your studio resources and your preferred style of working. Obviously, there are no generic rules – just what works for you.

-Kim.

2010/08/02 - 5 remix ideas for an 'a cappella' vocal

An 'a cappella' is, put simply, singing without instruments (or a backing track). When remixing, often an a cappella track is provided to the remixer. This is usually a track consisting only of the vocals from the original song, minus all the other instruments. It's also incorrectly spelled as 'acapella', 'a capella', 'acapela', or any number of variations.

Oh, ok. You have an a cappella track to remix. Or maybe you're working on a song and it's just not working. Or maybe you just want to have some fun. There are many ways of taking an a cappella recording and transforming it into a new song.

- Get active with pitch changing and timestretching. Modern pitchshifting and timestretching algorithms are excellent and, in a lot of cases, transparent in sound. This opens up a whole world of creative processing. You can change the melody or timing in a natural way and make as if the song was always sung that way. You could make extreme changes (or use inappropriate settings) and make the changes audible – making the processing part of the character of the sound.
- Find a new hook. Approach the lyrics and performance as raw songwriting material. Pick out a different line or section and use it as a hook or chorus. Rearrange the lyrics to give the song a different angle, or a different twist. Doing this you can create a new song that's based on the old one, but stands on its own – not in the original's shadow.
- Chop it up into little pieces and throw it to the winds (granular synthesis). Use the vocal recording as sonic material to be transformed into something completely different. This applies to any extreme processing. Create new sonic textures or other musical material that isn't melodically related to the original performance. This can be particularly powerful if the processing retains some character of the original sound – so that the resulting sound is familiar but not recognisable.
- Change the time signature. This can be a good idea if you're feeling low on inspiration or direction. If the original is 4/4, recast it as 3/4. Or 6/8. Or 5/4 if you're feeling daring! Taking this approach will instantly open up new musical ideas and provoke you to create something inventive.
- Record another performance of it. This is verging away from 'remix' territory into 'cover' territory, but can be particularly effective if the new performance is combined with sonic elements of the original song – either original vocal snippets or other sounds and instruments. Interesting things can happen if you blend the boundaries between remixes and covers…

These ideas should kick start a few of your own ideas. Experiment! Be creative! Whatever you do, don't simply add a dance kick drum and call it a day! There's a whole musical world of transformations and recomposition – go exploring!

-Kim.

2010/08/06 - Understanding Practice

(With apologies to Joe, who's actually a pretty cool guy)

Learn to smoothly and seamlessly practice your parts into that polished performance you're looking for.

You work so hard on your songs – spending hours tracking, mixing and mastering your songs. What's missing? **PRACTICE**.

Your mixes sound pretty good, but the timing just isn't as tight as you wish it was. You'd like you fix some of these issues, but you're not sure how.

Why would I want to practice my parts?

As a musician (or producer, if you're directing musicians) you're creating a performance. Your job is to make the performance sound as good as you possibly can.

As an engineer, you're capturing a performance. Your job is to capture it as truthfully (and as flatteringly, har har) as you possibly can.

As a producer directing musicians, unrehearsed musicians play out of time occasionally. When that happens, you have three options:

1. You can accept a mediocre performance.
2. You can educate your musicians about the wonders and joys of learning to play their instruments.
3. You can get better musicians.

Some people have a real issue with using practice to improve performances. They think it's hard work. **I agree.** It's really freakin' hard. And that's why it'll set you apart from the thousands of mediocre musicians who resort to technology to make up for their weak performance skills.

I'm a musician as well (no, really, I am), and I'm not ashamed to say I'm actually a really good musician. Two decades of music will do that to you. That said, I regularly make mistakes when I don't practice. Sometimes I can record a part over and over and over, and the timing is still a little off.

What do I do? I step away from the record button and actually practice the part. I focus on the difficult sections and train my fingers to work with precision and expression. Do I think it's cheating? Absolutely. I'm making myself a better musician through hard work. Using technology to fix bad performances will still result in bad-sounding music. However, practising a part to eventually put on a good performance makes sense. I view practice as helping the tracks sound as I intended for them to sound.

Ok, seriously…

Practice is really important. More important than you think it is. And you think it's pretty important. Well, it's more important than that. I'll go into more detail about how to practice effectively in a future blog post.

-Kim.

2010/08/09 - It’s not ‘amateur’, it’s just undeveloped

Sometimes I work with amateur artists.

No, it’s fine. It really is. They’ve often got some very interesting ideas about music (in a good way). Their fresh approach is fun.

Obviously, there are the downsides – less experience in performance, limited music theory knowledge, weaker work ethic, unrealistic expectations, etc. These are the kinds of aspects that separate a amateur artist from a more experienced artist. But this post is not about the artists. It’s about the music.

Music is different. The difference between amateur music and more refined’ music is production. Not sound quality – but old-fashioned production. Artist development. Compositional analysis. Creative direction. When I hear music from an amateur artist, it’s usually quite obvious. It’s not the sound quality that gives it away, it’s the music itself. The difference is in the crafting of the song.

An amateur song usually sounds like a collection of good ideas. Sometimes arranged in a structure that makes sense.

A more refined song sounds like a piece of work. Well-crafted. Integrated. Clear creative direction and execution. With character. With finesse. This kind of song has every second of sound – every note – deliberately supporting the creative direction, with not excess fat.

This difference is development. Amateur songs are usually called ‘finished’ when all the musical ideas have been arranged into some kind of structure. To rise to the next level, the song must be developed and refined. This is where a good producer adds value, and this is a large part of the work I do.

Often amateur music requires more focus and coherence with regard to main themes. Development and momentum are lacking. There’s not enough contrast or dynamics. The contour is not clear. The balance between stability and instability can be improved.

It’s not that amateur artists are ‘bad composers’. It takes skill and experience to know how to hear (let alone control) all these different aspects of composition. They’re subtle – more subtle than mix engineering in many cases. But they are as subtle as they are powerful. And it’s this difference that sets the more refined music apart.

-Kim.

2010/08/16 - Does your song need a hook?

Does your song have a hook?

Does it need a hook? Why / why not?

What is a hook?

A hook is a short musical motif that’s easily remembered and unique to the song.

As a short musical motif, a hook can be almost anything. It can be a distinctive melody. It can be an interested or arresting lyric. It can also be an instrumental phrase. Or even a silly sound. Any kind of short musical motif can be a hook.

A hook must also be easily remembered. In compositional terms, this starts with verbatim repetition throughout the song. Development should be minimised – the listener will learn to recognise it better if it appears in exactly the same form every time it’s heard. Better yet, having the hook appear at the same sequence point (for example, always directly following the chorus) will help establish its role in the song. This gives it a clearer identity, which helps with recognition.

A hook must also be unique to the song. The hook forms a big part of the song's identity. At best, it's used by the listener as a 'handle' for the song – the defining element that triggers the listener's memory. For this to be effective, the hook must be unique to the song. It can't be obviously similar to another song's hook.

Does your song need a hook?

Not every song needs a hook. Many well-formed songs don't have any element that functions as a strong hook. For some music, it's entirely appropriate for there to be no hook. A lot of film music fall into this category. It is primarily textural, and composed to support the visuals. A film cue may even be dramatic and vocal without needing a hook. Longer atmospheric songs may also function very well without a hook. For these songs, their defining characteristic is the sonic texture itself and the dramatic (or otherwise) contours.

Whether your song needs a hook or not depends on how you want listeners to remember it. If it needs to cut through the radio or iPod playlist, or if you want people to be humming it for hours or days after their listen, the song should have a hook. On the other hand, if none of these things are important, you don't need to worry about it.

-Kim.

2010/08/23 - Don't make better mixes. Make better music.

Stop it.

No really, **stop it.**

Stop focussing on the mix. There are more important things to focus on. Your mix is fine anyway.

Newsflash: The mix isn't really that important. Sure, a good mix helps the listener understand and enjoy the song. But a good song is still a good song regardless of the mix. And a terrible song is still terrible no matter how good the mix is. And listeners can tell the difference.

Making music is not all about mixing. Don't hide behind the technology. There's so much more to do:

Creative direction

Don't work on your mix. Work on your creative direction.

- Come up with new ideas for projects. Don't just work on individual songs – embark on something big. Something ambitious. Compose three songs (a demo) or six songs (an EP) to express a musical idea or feeling.
- Invent! Try out a new idea. Take a different direction to what you normally do. Invent something new. Combine two or more unlikely musical elements.

Composition

Don't work on your mix. Work on your composition skills.

- Focus on what the listener will hear. No-ones going to care how analogue your kick drum sounds if your song is boring and uninspired. It doesn't matter how 'tape-like' your mix bus chain sounds if the singer sounds like she just woke up.
- Melodies. Doing interesting things with sound doesn't count for much unless you're also doing interesting things with the notes. Learn how to compose an interesting melody. Don't just read a few online articles – practice! If you spent half the time writing melodies as you

do reading internet guff about plugins, you'd be coming up with beguiling and captivating melodies without trying.
- Harmonies. Don't just choose four chords and repeat them forever. Develop them. Let them grow and evolve throughout the song. Extend the chords. Use substitutions. Use slash chords. Vary the pace for dramatic effect.
- Rhythm. Enough four-on-the-floor. Disco is over. DJs have enough music to last until the sun spectacularly devours us all. *Do something interesting*. That's not enough. Now do it in 7/8. Alternate between 6/4 and 4/4. Seamlessly move between straight and shuffle. Vary the amount of syncopation for dramatic effect. Use composite time signatures. Juxtapose them. Some of the suggestions are silly, some aren't – and you won't know the difference until you try them.

Preproduction

Don't work on your mix. Work on preproduction.

- How will it all fit together? Think about all the elements in your song and reflect on what value they're adding to the song. Be clear about the creative direction of the song and cut out anything that doesn't support it. Have the courage to throw away good ideas.
- How can it be strengthened/improved? Don't stop when you have all your material arranged into a structure that makes sense. How can you continue to improve the music? Pay attention to the contour of each section. Make sure each transition (from one section to another) is clear and deliberate – not just one section after another.

Rehearsal

Don't work on your mix. Rehearse your parts.

- Physical instruments. Yes, I know we've got Elastic Audio and Autotune and endless disk space for multiple takes… but they're no substitute for a good performance. Editing can turn a sloppy or lazy performance into a competent one. It can't, however, turn even a competent performance into an inspired one. Editing can't add expression or feeling or excitement to a performance. No technology can – it comes from the performer. And the performer can only do it after hours of practice and honing the craft. So get on it.
- Virtual instruments. Oh, you thought virtual instruments are different to physical instruments? Take a look at those black and white keys under your fingers. Take a look at those assignable knobs. Make a performance of it. Put some expression into it.

Oh yeah. That sounds like a lot of work. Making music is a lot of work. Cry me a river. You think you can become successful by being lazy? Yes, you *do* want to be successful. Success isn't a record label contract or a sold-out stadium. Success is honing your craft. Success is becoming insanely good at what you do. Success is shipping.

And if you see yourself exclusively as a mix engineer? Make sure your clients do all the above.

-Kim.

2010/08/26 - Come into my Kitchen…

Some of you who have been paying close attention may have noticed some references to a project called 'The Kitchen'.

Those who have subscribed to my email list (there's a form at the top right of this page) were given the full details about this a couple of months ago.

The rest of you might at this point be wondering – **What is the Kitchen?**

Essentially, the Kitchen is a way for me to help you with your music projects. My help. Your music. You and me. Detailed and specific assistance. It's a little bit like hiring me as your producer. Except it's online. And it costs less (more on that later). And it's a small community of friends.

Think of it as:

- A private community that is serious about making better music.
- A way for me to give you direct, personal assistance on your projects.
- A place to discuss music making without trolls, distractions or bad advice.
- A small group of trusted friends, not a public forum.

What are the possibilities?

- Having difficulty mixing a tricky song? **I can mix it and tell you what I did.**
- Feeling confused about something? **I can write a detailed tutorial just for you.**
- Having trouble writing a bassline or chord progression? **I can write parts *with* you.**
- I can give you feedback on your work at multiple stages – composing, mixing, mastering, etc.

And that's not all… There are some PDF documents at the Kitchen as well. Some of you have asked for a downloadable (and printable) PDF version of this blog. The PDFs at the Kitchen are better! They're based on some blog posts, but they're better-written and integrate concepts together to give you a better understanding of how different techniques work together.

The Kitchen has been running privately for several months and over a dozen people have found it valuable.

How much does it cost?

This is the internet in 2010**.** You are from all over the world, with different capacities to afford this kind of help. You might have fluctuating needs – bunkering down to work alone one month, followed by intense co-writing or co-mixing with me the next month. I might provide help every few days, or you might be happy to read the PDFs and casually chat about the music industry. Everyone's needs are different.

The Kitchen works on a monthly 'pay-what-you-want' model. This means the Kitchen is as affordable or valuable as you want it to be. That means you're not tied to a yearly membership. That means you can dip your toe in the water and get a taste if you want.

If you want to be part of it next month (and I hope you do!), simply follow the link below to make your payment. Further details will be provided by email.

https://www.paypal.com/cgi-bin/webscr?cmd=_s-xclick&hosted_button_id=SMS2L5QH5UNWA

Signups are now open for September. See you at the Kitchen!

-Kim.