Signal Processing In The Analog vs Digital Domain: Which Is Favourable For Mixing And Mastering Music

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Throughout recent years, the music industry has shifted how people listen to and create music. The move has been away from analog - using circuitry alongside a variety of other physical techniques - to using a fully digital approach, from recording to storage. The vinyl record and cassette tape have seen a vast decline in their use for music playback while digital consumption has risen exponentially. This essay will discuss the distinctions between analog and digital with regards to their role in music processing in the mixing and mastering stages of production, attempting to suggest when one may.

Mixing is commonly defined as the process of gelling all individual elements of a track together to make it sound coherent, balanced and levelled across the frequency and stereo spectrums. The stereo spectrum is commonly defined as where the element is perceived to sit in space. A mixing engineer tweaks the stems to allow them to run alongside each other and have their own space in the mix.

Mastering involves the final treatment and enrichment of the piece of music in its entirety. It entails manipulating the volume to ensure it is acceptable for commercial levels and flows smoothly with the tracks either side of it in an album, if applicable. A mastering engineer may also enhance certain areas of the frequency spectrum or use other techniques to apply a certain tone to the entire track. It is the final process before official release.

There exists an array of effects commonly used in these processes: equalisation by cutting or boosting specific frequency ranges; saturation/distortion to thicken and add colour to an input; compression to attenuate the loudest parts of a sound or limiting to completely eliminate them; and reverb to add space and ambience by replicating a given acoustic environment.

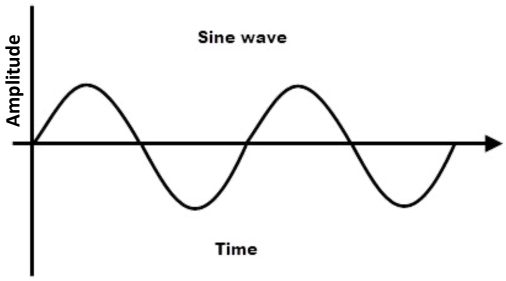
These have all existed in analog form for years, and as the digital revolution has strengthened its grip they have been converted into computerised versions. This change considerably increases ease of access, user interface and provides several other benefits. It is, however, important to recognise ways in which using analog equipment to edit music is more suited to certain tasks. Its physical nature makes it susceptible to multiple phenomena which affect the way the sound is processed and consequently heard. Because digital effects are simply coded algorithms which can perfectly change a sound in the exact same way every time, they are flawless in their output and limitless (if the right code is used.) Analog gear is finite and can only handle voltage up to a specific level before it begins to falter technically, and the sound is consequently distorted in some manner.

In some regards this can provide an issue for the engineer; distorted sound may be unwanted and damage the quality of the pre-processed sound. Yet, in some cases the analog “fuzz” is a colour to be strived for, which many modern digital software engineers aim to replicate but struggle. This essay will discuss the relative advantages of analog and digital processing, based upon their ease of use, consistency but most significantly comparing effect on the input sound.

Saturation:

Saturation is undoubtedly one of the most vital effects needed for a thick coherent sound which is present. It involves changing and manipulating the waveform and thus altering the frequency content, adding harmonics to the sound. Harmonics are whole number multiples of the fundamental frequency, which is commonly defined as the frequency at which the entire waveform oscillates.

All saturation is based off “driving” a soundwave through a saturation curve. Let us define some digital soundwave as a relationship between time and amplitude, demonstrated by the sine wave below.



The sine wave is a simple model to use as, owing to its shape, it is perceived as a pure tone, or a single frequency oscillating without any overtones (additional frequencies above it.)

FIGURE 1 - SINE WAVE AS AMPLITUDE-TIME RELATIONSHIP (ELECTRONICS HUB, N.D.)

A saturation curve shows a relationship between input (x-axis) and output (y-axis) amplitude, thus as the sine wave is driven through the saturation curve it outputs a different amplitude at each point in time, thus producing a new wave.

A change in the shape of a digitally represented soundwave suggests a change in the frequency makeup of said wave. The Fourier transform demonstrates that any complex waveform can be represented in its simplest form as a sum of sinusoidal (taking the shape of a sine wave) waves of varying phase and amplitude. Therefore, if the shape of the overall waveform is changed it will be broken down into a different set of waves. Certain frequencies will be emphasised more/less, and some new ones added/removed, for the sum of them to give some new shaped wave. Let us now look at an example in the analog format.

Analog saturation can occur due to a wide range of causes. Soft clipping is a common result of processing an audio signal through an analog chain or piece of equipment. Instead of an amplitude-time relationship, sound is processed as a varying voltage over time in analog equipment. Therefore, it can be edited by passing it through transistors, capacitors and a variety of other circuitry to change the sound when it is reconverted to audible sound. When a voltage input exceeds that which a piece of equipment is able to handle, it is cut off or squared off in a non-linear manner i.e. any increase in input voltage will have disproportionate/no effect on output voltage. The wave gradually eases into non-linearity from its linear section, henceforth why it is referred to as soft. Hard clipping (when linear becomes non-linear at an instant) can also occur, but gets rid of bass frequencies and adds harshness so tends to be avoided.

If a wave has been subject to soft clipping in analog gear, its shape has thus changed as the peak voltages - both positive and negative - have been brought down to a lower magnitude, having been rounded off. As discussed previously, the change in shape of the waveform adds harmonics to the sound. This also provides a gentle compression to the sound as the loudest peaks are being attenuated, reducing the dynamic range.

To greater understand the effect of different shape changes on the sound and the nature of the harmonics added, it is necessary to look at saturation curves in more depth which is far more easily done with digital processing. Let us first take a sine wave at frequency 440 Hz, the note A4 in musical terms. For the purpose of visualisation, a spectrum analyser will be used which shows the relative amplitudes of each frequency in a sound, with frequency on the x-axis and amplitude on the y-axis. It uses a fast fourier transform to convert the complex waveform from the time-amplitude domain to the frequency-amplitude domain to gain a better understanding of frequencies present in a sound.

We observe the sine wave is a single peak at 440 Hz since it is a pure tone as discussed earlier.

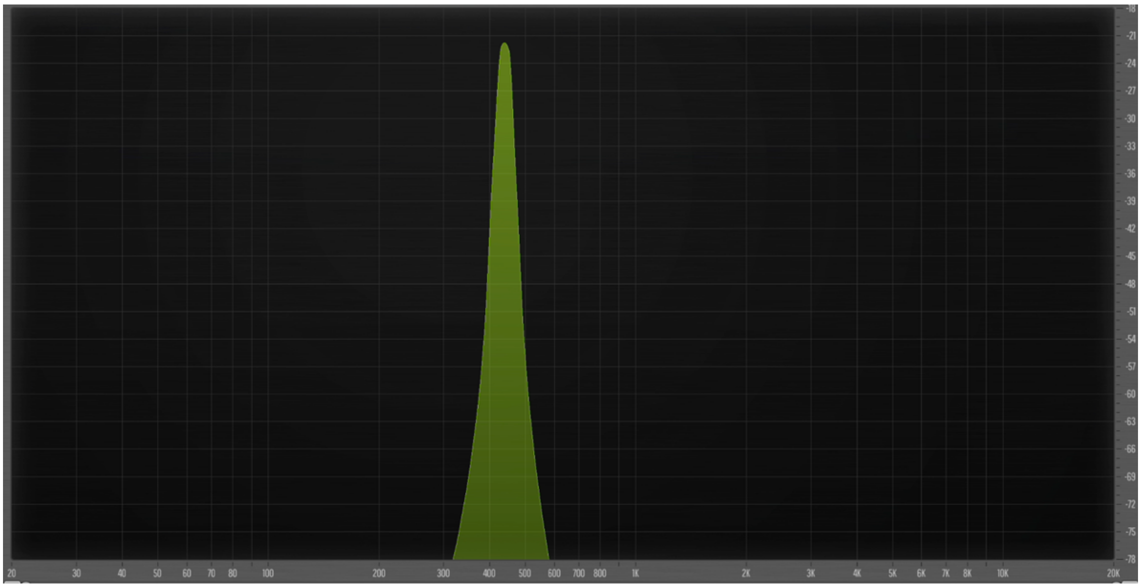


FIGURE 2 - SINE WAVE AT 440 HZ SHOWN ON SPECTRUM ANALYSER

Let us take a simple saturation curve, such as the one corresponding to the soft clipping concept discussed above. We can notice how it is linear in nature up until a certain point at which it begins to round off, just as the effect of soft clipping occurs.

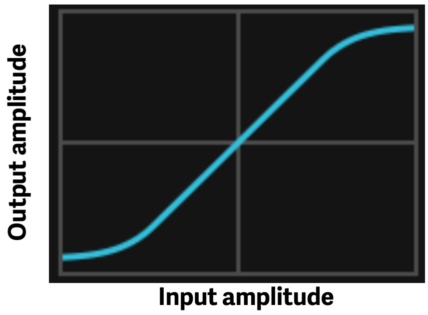


FIGURE 3 - GENTLE CLIPPING SATURATION CURVE

As we begin to drive the sine wave harder through this curve, it begins to peak off at a lower amplitude and thus changes shape slightly. This variation will not change the phase of the wave so the fundamental frequency remains unchanged, whereas the overtones have. We can see these overtones on the spectrum analyser below.

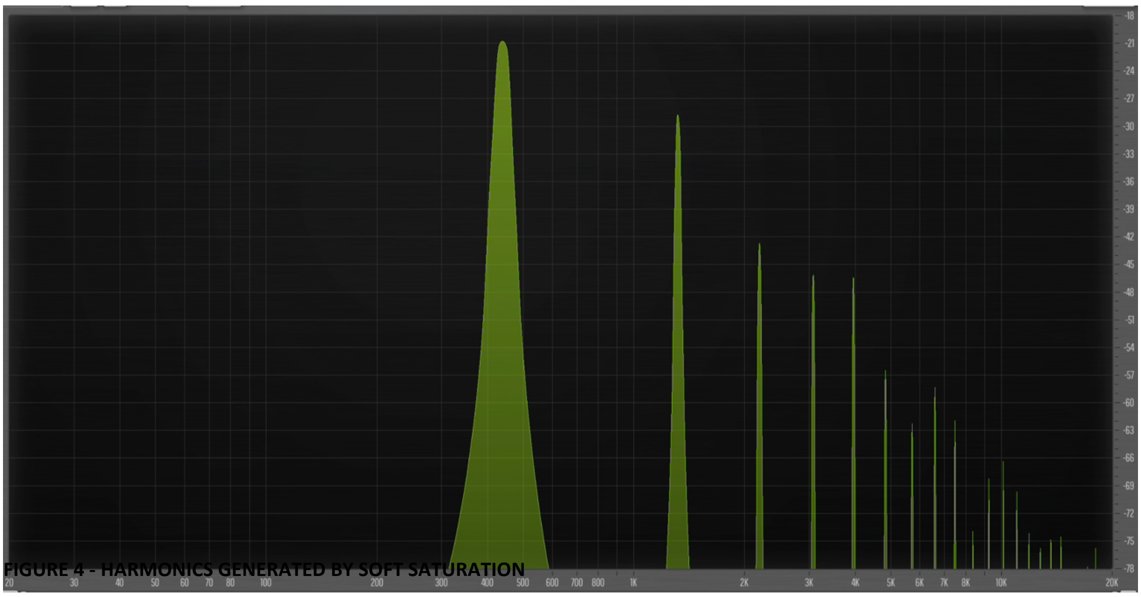


FIGURE 4 - HARMONICS GENERATED BY SOFT SATURATION

As we can see, the fundamental frequency remains present but there are now also peaks at 1.32 KHz, 2.2 KHz, 3.08 KHz on the spectrum. These are odd whole number multiples of the original 440 Hz (3*440 = 1,320 etc.,) otherwise referred to as odd harmonics. The amplitude at any point beyond the clipping threshold on the sinewave has changed, meaning the shape has also, and consequently overtones added.

The saturation curve is described as "odd symmetrical" as the relationship is identical but in opposite quadrants i.e. when rotated by pi about the origin is equal to itself. Odd symmetrical curves give rise to odd harmonics as displayed above. Even symmetrical curves (i.e. those which are reflections of each other in the y-axis) give only even order harmonics and remove the fundamental frequency, since it is an odd harmonic (1*fundamental frequency.)

The ease of manipulation provided to an engineer by digital forms of saturation is undoubtedly useful when it comes to ensuring the elements of a track synergise together. It provides total control over all parameters such as the output of a saturator (since giving more presence to new tones will increase loudness and this may need to be accounted for by reducing output volume) and of course, the specific curve through which the waveform is driven. Since it simply works through a set of code, it produces a predictable and repeatable output which guarantees a desired sound. For example, if an engineer had two sounds clashing in a similar frequency range they may saturate one with odd and the other with even harmonics leading their corresponding overtones to intertwine together across the spectrum, despite the fundamental frequencies still potentially clashing.

Despite its relatively unpredictable nature at times, analog saturation from soft clipping also has advantages in its sound. It has become known as a component of the "analog warmth" sound which is present on most records mixed and mastered in the physical domain. The interplay of all different components in an analog processing chain adding their tone to the sound from gentle clipping of the peak voltages and overheating of the gear causing slight changes in the waveform passed through it, result in a very specific colour. The added, often slightly random in nature, harmonics fill up space in the frequency spectrum where specific constituents of a track may not, thus giving the overall mix a beautiful fullness. This is something which software engineers have sought to replicate in digital form and which has been achieved to some extent, but many engineers continue to work in analog solely to maintain this quality to their work.

With the imperfection of the waveform distortion from analog processing being such a desired concept across many genres of music, it provides a recognisable advantage over the perfect, algorithm based digital world for some engineers, making it a worthy reason to support the continued use of analog processing.

Compression:

Compression involves the attenuation of the loudest parts of a sound, in order to make its level (volume) more consistent throughout the track. A threshold is usually set by the user and any volume peak above this is turned down by a set ratio. For example, if a 4:1 ratio was set, an input of 4dB above the threshold would result in a 1dB output. This aids in reducing the dynamic range of an instrument or sound by bringing the peaks closer to the troughs of the level, which is vital in ensuring it sits at a consistent level in the mix. If there are notably loud peaks and softer quiet sections, the element of the track in question fluctuates between being inaudible and unwantedly dominating the mix.

The other two main compression parameters are the attack and release times which can be manipulated to change the shape of the sound. Attack refers to the speed at which the compressor acts, so a short attack time would result in the compressor acting very quickly and clamping down on the loud, punchy start of the sound – the "transient." The transient is the peak at the start of the sound followed by the sustain. The shaping of transients plays a large part in drum processing as it determines the . A longer attack would cause the compression to be delayed and not take effect until after

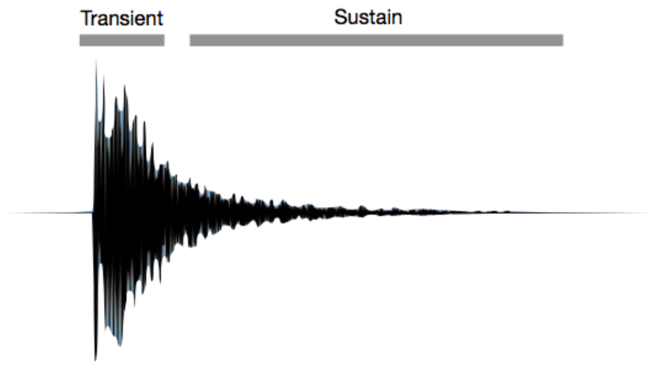


FIGURE 5 - LABELLED TRANSIENT OF SNARE WAVEFORM (SPLICE, 2017)

Release can be commonly defined as the opposite of attack; the time it takes after the above-threshold input stops being received for the compression to stop acting. This can also be utilised to have the compression squash the entire note, just the transient, or even the following note/hit.

Analog compressors encompass a variety of different circuits, with one of the most common being the Voltage Controlled Amplifier (VCA) compressor. They work by receiving an input voltage signal and first detecting the extent to which the voltage is exceeding the threshold set by the user. When this disparity is determined, there is a "control voltage" produced. The more the input exceeds the threshold, the greater the magnitude of the control voltage produced. This is then adjusted by a potentiometer according to what the ratio is set at, so will increase the control voltage more as the ratio is turned up from, for example, 2:1 to 10:1. A VCA then receives the control voltage and reduces the gain (amplitude) of the signal at that point and for the duration it is receiving some control voltage.

Analog compressors may be favoured for multiple reasons. Since they work off circuitry, their curbing of audio peaks can be more natural sounding. Referring to previous ideas of non-linearity, compressors input will tend to fluctuate in voltage quite significantly and contain some large peaks which the compressor may not be able to handle. If the circuit components such as the transistors and VCAs involved are overloaded, they will begin to lightly distort the sound by changing the shape of the input wave and adding the analog warmth.

Indeed, many specific compressors are also desired because of their distinct tones heard on famous records. The 1176 by Universal Audio is recognised for being ahead of its time with its groundbreakingly fast attack (20 µS) and release (50 µS) times which can be heard on iconic vocalists, from Michael Jackson to the Beastie Boys. (Universal Audio, n.d.) The sound has now become so distinct across engineer communities because of the heat it adds to recordings.

Digital compressors are far more precise. They simply take the input and complete the task according to the parameters given. There is an array of advantages to using these over analog gear. Visual representation of the parameters and corresponding gain reduction is a significant one one, allowing an engineer to tweak accordingly and with excessive precision. Another useful feature is the continuous nature of parameter on digital compressors, as opposed to discrete on analog compressors. These will often have a few set ratios to be selected from: 2:1, 4:1 and 10:1 being the most common. Digital equivalents provide a range of values, in between which any value can be taken allowing for more control over the sound.

Limiting:

Limiting is an extreme form of compression, where the ratio is either very high or infinite, meaning any peaks above a threshold are completely squashed and eradicated. Limiting is crucial, especially in the mastering process to ensure a track does not break commercial and legally acceptable volume levels. Engineers also use limiting when mastering the entire song to increase perceived loudness. By cutting off all peaks above a certain point, they are able to drive the track louder into the limiter, thereby increasing the volume of the quietest part of the music without having any affect on the peaks.

Again, digital limiters offer far more precision and flexibility in their function. As limiters are so key in their role of ensuring music, podcast and TV audio comply to suitable levels, having full control over the limiters threshold, attack and release times and input gain is extremely useful. Technology allows for easy use of multiband limiters, which place limiter constraints or different limiter parameter values on different ranges in the frequency spectrum. They also allow for pristine outputs with no change to the sonic quality of the input, just simply affecting the levels when triggered.

The arguments for analog limiters reside predominantly in the colour and quality they bring to the sound. The "mix bus" is the bus (channel) through which the entire mix flows before being played back. Running this through a sonically colourful limiter can aid in limiting and gluing tracks together to make the mix sound coherent, applying the same harmonic distortion and heat to the entire piece.

Slew rate is commonly defined as the "maximum rate of change of output voltage per unit time." (Electrical4u, n.d.) Often, analog components which edit the voltage of a signal can be subject to "slew rate limiting" when the ΔV_{output} is too great across a time period for the component to respond, so it naturally decreases the gradient of the change in voltage. This results in a subtle distortion of the signal in multiple ways. It can have a similar soft clipping effect to discussed earlier, as the transients are not able to be reproduced in their entirety due to the slew rate cap and are rounded off slightly instead.

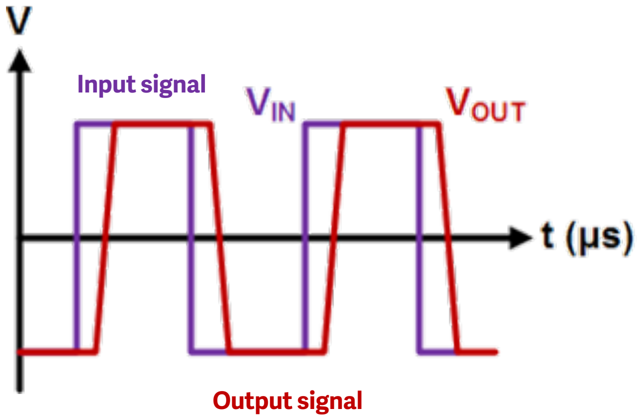


FIGURE 6 - EFFECT OF SLEW RATE LIMITING ON SHAPE OF VOLTAGE SIGNAL (MICROCHIP, N.D.)

An alternative issue which may be introduced is phase-based issues. Waves of the same frequency can be "in phase" when they are aligned i.e. their peaks and troughs occur at the same point in time. When gear is unable to respond to sharp changes in signal voltage and is forced to make the change over a longer period of time, it can push the signal slightly out of phase. This has the potential to reduce the clarity of the sound or entire track if we hear a version of it slightly after we expect to and makes the overall mix muddier as frequencies are beginning to sit in unwanted positions in time. This is an issue to be considered when using analog equipment, and can be completely avoided through the used of software to process music.

Accordingly, the preference between analog and digital for both limiters and compressors is determined by how much an engineer values precision and accuracy against timbre and colour. Given how important levelling and loudness manipulation is in mixing and mastering to ensure each element has its own space without being too overpowering, getting compression correct is necessary. Drum processing revolves heavily around transient handling, as a punchy and present transient is frequently the main attraction of any drum samples in comparison to other instruments which are pitch based/tonal and smoother. Therefore, whilst the vibrance provided by outboard analog gear is striking, complete control over a wider variety of continuous parameters means digital may be the more suitable option for compressors and limiters, especially given their multiband capabilities i.e. different parameter settings for different frequency bands which can open up a new realm of creativity.

Reverb:

Reverb is a more creative effect when compared to the more technical ones previously discussed, but still has its place in the post-recording processes. It aims to imitate some acoustic atmosphere, from a room to a cathedral, and what the input signal would sound like when played in this environment. The notion originates from lots of fast reflections of the soundwave travelling in all directions around the room and assimilating into one perceived sound. This gives a listener in the space the impression the sound is coming from all around them (varying depending on the size, shape and acoustics of the room.)

Reverb is often used in parallel processing. An identical version of the signal is sent from the original track to a "bus" (channelling route) where it can be further processed without affecting the original. Reverb is then applied to the separate bus instead of the original track to maintain the dry (unaltered) signal alongside the wet reverb signal. This way, the listener hears the original recording and the reverb signal as if they were in the room, giving a far more realistic aural experience. It also offers more control over the reverb itself – the engineer can control the amplitude of the signal sent to the reverb bus and process the output from the reverb through further chains without affecting the original.

Reverb is a challenging concept to emulate using circuitry, thus there are a variety of ways typically used to do so. Plate reverb is among the most common. A thin sheet, or plate, of metal is placed in the centre of a box in a frame to keep it still and put under tension to ensure uniformity throughout. An input transducer converts an incoming electrical signal to a physical audio signal which passes through the air in the box and causes shimmery vibrations in the sheet. This is picked up by one or more pickups, or output transducers, which gather the vibrations specifically in the sheet - not the surrounding environment - and re-converts them back into an analog signal. This adds a sense of dimension and width to the sound.

This shape of this reverb signal can be altered depending on the characteristics of the plate. Decay time is typically defined as "the time (in seconds) required for the reflections (reverberation) to die away," (Presonus, n.d.) and alongside pre-delay (time taken between initial dry signal and first noticeable reflections) can be manipulated to change the perceived size of the room and position in said room providing the reverb. A larger plate under less tension means the vibrations through it linger for longer, thus a longer decay time replicating a grander setting, like a cathedral. This is because the more tension in the plate, the higher its resonance frequency ("the natural frequency of an object where it tends to vibrate at a higher amplitude" (Cadence, n.d.) when excited by the input transducer.) Thus, the higher frequency vibrations in the plate dissipate their energy through heat and sound to the surroundings quicker, mimicking a quicker decay time. The opposite is also true: lower tension gives a longer decay time, representing reflections taking a longer time to travel around an environment indicating a larger room. However, the difficulty involved in editing these parameters can make them less attractive in comparison to easy-to-edit digital plugins.

Digital reverbs vary slightly in their details, but generally follow similar structures with regards to their parameters. They often come with presets which are named and grouped according to the environment they attempt to emulate i.e. a small drum room, but also often seek to copy analog reverbs, namely the plate reverb effect.

Digital reverbs first put the sound through multiple delay lines, which are mathematical algorithms whose role is to delay the playback of the input signal as though they have travelled, hit a wall, and proceeded to bounce back. Sounds which have only hit one surface and returned to the listener are referred to as early reflections, and as these continue to redirect off other surfaces/pass through more delay lines they become late reflections. The relationship between late and early reflections can create an extremely vivid aural perception of where a sound is coming from in a mix. Engineers delay early reflections to make an element seem to be far away in front of the listener in a room. Alternatively, loud and abrupt early reflections suggest the instrument or sound is right in front of the listener, helping to achieve the sensation the listener is “inside the track” when combined with manipulation of the stereo (left/right) spectrum.

One reason for the appeal of digital reverbs is the development of convolution reverbs. These can almost flawlessly produce the outcome when an inputted sound is played in some acoustic environment by analysing and copying the process when every frequency in the human audible spectrum (circa 20Hz – 20,000 KHz) is played in that environment. The effect of the space is logged by collecting an impulse response (IR) in that physical environment. This is an abrupt, atonal sound which spans a large portion of the frequency spectrum which is played out to record the behaviour of the reverberations, which will differ at each frequency due to the varying energy of the different vibrations. The dry sound is removed, and then any input can be passed through the IR in the reverb software to take on the quality of being heard in that space. The reverb convolutes the sounds together through a complex process, essentially playing the input sound through another audio file.

Again, the upper hand here depends predominantly on an engineer’s creative choices. Digital plugins can play sounds in an unfathomable range of acoustic environments which each treat the sound differently. Some frequencies are absorbed slightly more in certain environments, while others are more resonant. It is possible to replicate this in plate systems by further processing the reverb bus, but computers enable easier use and broader experimentation. Moreover, an essential aspect of mixing is the paths which the signal of the tracks follow. They may be channelled/processed through a bus alongside other similar instruments to be treated with the same effects, or multiple tracks “sent” to the same reverb track, for example. The signal does not simply get treated on its own track and then played out of the master output straight away. Mixing in digital, therefore, enable the engineer to explore far more complex pathways, as there are essentially unlimited tracks to use for routing and grouping sounds and instruments together. There are many ways to add reverb, in series i.e. directly onto an instrument track or in parallel to maintain the dry signal and with varying frequency contents, each giving rise to a different comprehensive sound. Thus, while the analog reverb effects may or may not be preferrable to digital reverbs due to their authentic and unique timbre, the ease of use and applying reverb in the digital domain allows creative decisions in the mixing process to be made fluently and easily.

Equalisation:

Equalisation, shortened usually to EQ, refers to the process of increasing or decreasing the gain (level) of a frequency or band of frequencies on a track and is arguably the most important effect in modern music. This allows each aspect in a song to be present in its own range of the frequency spectrum without interfering or being outdone by another. Without it, every commercially released piece of music would sound muddy, harsh and incoherent.

EQ is an extremely broad term and everything including the parameters will vary depending on the type, but most come in both digital and analog form, with both having their respective disadvantages and advantages. There are thousands of digital replications of classic analog gear which attempt to recreate the pleasant nonlinearities heard which grow ever more accurate, although there are some ways in which analog equalisers make themselves the obvious choice.

One of the most frequently occurring types of EQ is the parametric EQ. It splits the frequency spectrum from 0-20 KHz into frequency bands, the number of which can vary from three upwards. The spectrum is represented in logs as the audible range involves “several increases in order of magnitude,” (Science Direct, n.d.) and when you go up in musical pitch by an octave, you double frequency thus frequency exponentially increases as you go up the conventional musical notes. Each band is symmetrical about a centre frequency (CF) which gets boosted or cut when the level is increased or decreased respectively. The extent to which the surrounding frequencies of the central experience the level change is determined by the quality factor, Q, which can be commonly defined as the CF/BW, where BW is the range between the points 3dB above or below the CF, depending on whether a cut or boost is occurring. For example, a -9dB cut is occurring at the 3 KHz band and the frequencies 1500 Hz and 6 KHz experience a -6dB cut. 3000/4500 = 0.667 = Q in this case. (Recording.org, n.d.) When it is present (which is uncommon in parametric EQs on analog mixing desks,) Q can be increased to make the bandwidth smaller and affect a smaller range of frequencies, as centre frequency is constant. The inverse is also true, meaning a band can span several KHz with a low Q.

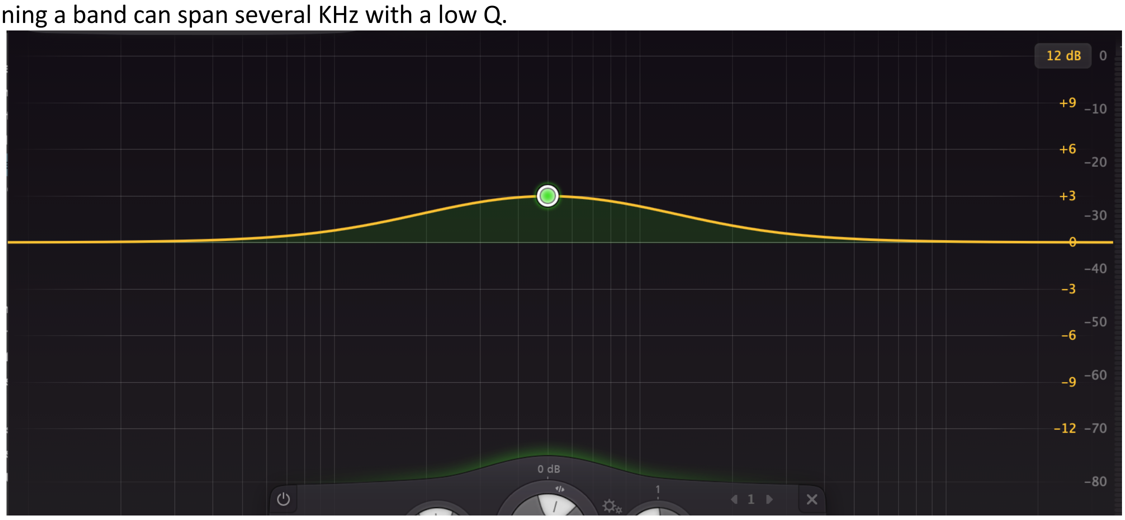


FIGURE 7 - BOOST OF 3DB AT 500 HZ WITH QUALITY FACTOR 0.5

FIGURE 8 - BOOST WITH IDENTICAL PARAMETERS APART FROM Q RAISED TO 3 FROM 0.5

EQs are often used to surgically remove jarring and harsh resonant frequencies in an instrument, such as the "s" sound in a vocal recording which reside in the 5-7 KHz range and are exceptionally grating to the ear. Alternatively, they are used to boost certain ranges to give colour i.e. the >7 KHz range to add "air" and "sparkle," or the 200-400 Hz range to add "body" to a sound. Many engineers consider that digital is generally better for cutting specific problem frequencies. Due to a digital EQs precision and control over the range being cut and its ability to locate and completely remove a specific frequency, it proves very useful in comparison to analog which tends to works in broader bandwidths. An engineer can isolate each frequency and listen up the spectrum to easily find the issue and turn it down by however much may be necessary by reducing the gain of this frequency.

Instead, many prefer analog equalisation gear in their processing chains for boosting ranges across the spectrum. An engineer may do this when mastering a track to add presence to a frequency range which may be weaker and inaudible. Digital EQs simply turn up the targeted frequencies – it is a mathematical increase in amplitude without adding anything extra to the sound. Analog EQs often have a specific sonic colour to them which digital replications seek to achieve, once again deriving from non-linearities in the physical circuitry when driven slightly. Unlike digital effects which tend to introduce harsh unwanted "artefacts" and distortion into the sound when driven to extremes, analog EQs smooth off the peaks of the now louder waveform, adding subtly to the sound as well as just making it louder.

Analog equalisers differ from digital ones in their lack of visual feedback. A computer user is often faced with a full frequency spectrum - as seen in figures 7 & 8 - which allows them to observe the feedback from changes which can detract from their ability to determine the best sound; they may focus too much on what they see. Analog EQs force an engineer to hear the effect they have as they tweak parameters, thus arguably leading to a better output. After all, they are pursuing what sounds best in their work so making decisions based solely on this can be advantageous. Although, the value of seeing the spectrum should not be underestimated. The point of an EQ is to "balance" the frequency spectrum, therefore where ears may fail to perceive unbalanced frequencies visual responses can provide a platform to do so. It begs the question of what is more important as the final product of a mix and final master: a fantastic sounding, soul-filled track which may be subject to gentle distortion or a sonically stable and coherent optimisation of sounds and instruments which may lack depth at times?

Regardless of one's approach, mixing and mastering are creative processes. Indeed, their aim is to convert an array of raw, dynamically inconsistent and often muddy stems into a consistent piece of music for commercial listening. But the choices made can yield thousands of different interpretations of the song. 90s hip-hop would not be the same without up-front, heavily compressed vocals. Modern house music would suffer without a tight low-end with a well synergised kick drum and bass which do not reside in the same narrow frequency range. Even commercially recorded classical music would sound unusual without EQ processing due to the frequency damping qualities of microphones.

Digital processing allows for almost infinite sound manipulation possibilities, extreme ease of change and experimentation. Any signal can be processed alongside any other, in parallel or series and with concise detail on parameter values to ensure a technically flawless mix. The control over placement in the stereo spectrum and perceived distance from the source can give the listener a sonic experience like never before when listening through a high-quality pair of headphones or speaker system, as this provides two sources separately for the left and right sides. Vision and experiment are enabled by digital sound production and editing.

Yet, it is notoriously difficult to either substitute or accurately emulate the changes to the signal caused by analog equipment's inability to process specific patterns in voltage. The inevitable gentle distortion to the sound from long analog chains provide a thickness and colour to an individual stem or entire record from the harmonics generated from the waveform changes. Moreover, interplay between different pieces of equipment which all add their individual sonic tint to the output has the potential to generate remarkable but distinctive tones to a track which become associated with certain engineers and their works.

Both methods come with their advantages and drawbacks purely in their effect on the sound. The extent to which each is used, often in combination, is determined by the desired final sound of the mixing and mastering engineers and original artists and producers. It is difficult to propose one as objectively more useful to the music industry than the other; both should and will continue to be utilised to provide the astonishing sonic range of modern commercial music.

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(I generated the uncited diagrams using Ableton Live’s Saturator, FabFilter Pro Q-3 EQ and Voxengo’s Spectrum Analyser)