RECORDING, MIXING & MASTERING TIPS

An introduction into recording, mixing & mastering (Final Version 3.0, Jul 2024) By DutchGuitarist

Content

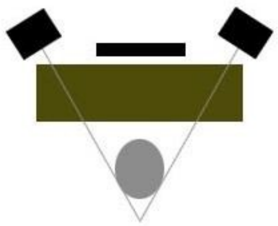
Introduction

This is not a recording tutorial. I just want to share my experience in this area and maybe it will help you a little bit. Apart from recording I will cover is the use Virtual Studio Technology (I'll explain later), use of effects such as compression, reverb, exciters etc and mixing and mastering tips. New in this version is the fact that is aimed at a 64 bit environment rather than the previous version that was aimed at 32 bit environment. This means that the plug-ins mentioned in this document are all available as 64 bit. If you have bridge software(such as Jbridge) you can still use the 32 bit plug-ins in a 64 bit environment.

General approach

I use effects sparsely and only enhance the track in such a way it sets better in the mix or to modify an otherwise dull track. Also I keep the mixed down output level under -6dB's for the mastering stage. My Digital Audio Workstation(DAW) is Cubase 11liniy, but the tips here also apply to other recording systems.

Monitor placement



This may seem obvious, but a common mistake is that monitors are not placed properly. Monitors need to be as far apart from each other as you are from them. So in other words the distance should be a perfect triangle. And of course the height should be on ear level.

Monitor Volume

As a rule of thumb: The volume of you monitors should be set that high, you can still have a conversation with someone nearby without having to raise your voice. When you listen at low to medium volumes, you tend to hear more midrange and less of the lows and highs.

Meanwhile there is also something called the Fletcher-Munson effect, which is about how different frequencies are heard depending on the playback volume. It is wise to check your mastered song on different volumes from time to time.

About Virtual Studio Technology

Virtual Studio Technology (VST) is a software interface from Steinberg that integrates software audio instruments and effect plug-ins with Digital Audio Workstations. VST uses digital signal processing to simulate traditional recording studio hardware in software. Thousands of plug-ins exist, both commercial and freeware. I use both the instrument as well as the effect plug-ins during the mixing and mastering stage. In the remainder of the document I will give examples of free 64 bit VST's I personally use.

Recording

Introduction

This section of the document is aimed at home recording and it contains tips and tricks I ran into during the recording stage.

General approach

The first and most important (and obvious) tip is: tune you guitar (or any other instrument for that matter) en check this after every recorded track. Too often changes in temperature or new strings on your guitar are causing slights out of tune sounds and we do not want to record this (normally that is) Also record at the highest possible resolution. A higher bit depth enables you to have a greater dynamic range and captures more subtle detail in your recordings.

So if possible record at 24-bit/96kHz. On the downside, a 24-bit file is 1.5 times the size of a 16-bit file. So you need some extra space on your hard disk.

Also remember that the CD norm remains 16-bit/ 44.1kHz or 48kHz. In the mastering section of this document I will explain how we can bring the 24-bit/96Khz back to CD standard using a technology called **dithering**.

Standardise your tracks

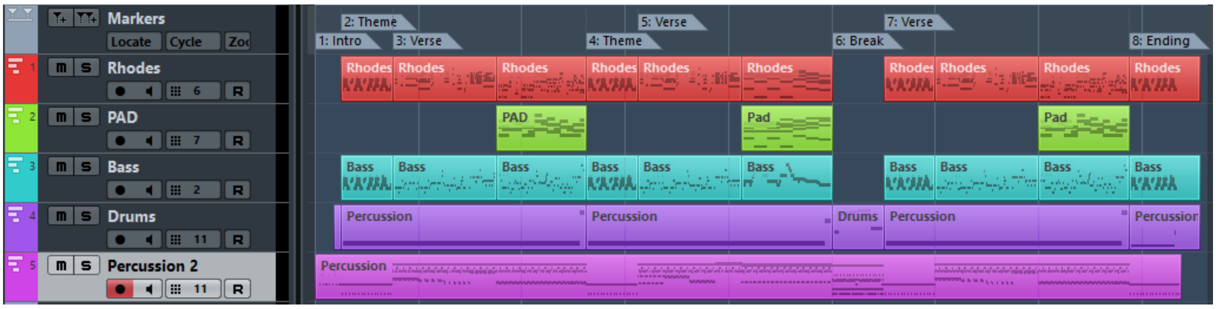
A time is a valuable asset, it is a good approach to standardize your tracks in two ways: the order (from to bottom), the colour of the tracks and the event descriptions (markers). I start with the chard and melody tracks at the top and then synths/strings section followed by the bass and finally drums. Recorded audio material is always at the bottom. It is also good habit to give the individual track specific colours as a visual aid to what type of track it is. Personally I have the following group scheme I (always) use (also notice the track order):

Keys: Red
Synth pads/strings: Green
Synth leads: Orange
Brass: Dark blue
Bass: Light Blue
Drums & Percussion: Purple

It does not matter what colour you use or what track order you use as long as you are consistent

If I use multiple track of the same type I vary the colour slightly. In the following example I use a drum and a percussion track:

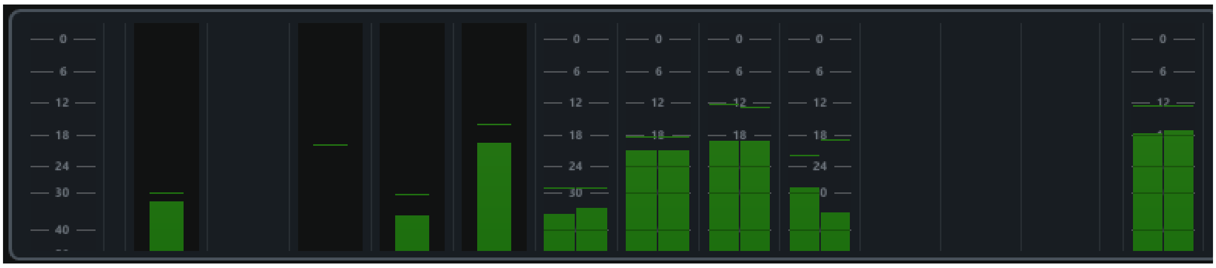
Finally , if you DAW allows it use markers to indicate special events in the recording. This way a project could look like this:



Recording levels

Record audio at levels in the −18 to −12 dB range on the meters. Unlike tape, digital recordings should **NEVER** clip. So keep a close look at the meter bridge and adjust in incoming signal when required.

Meterbridge example:

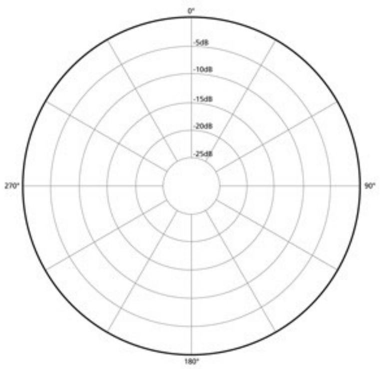


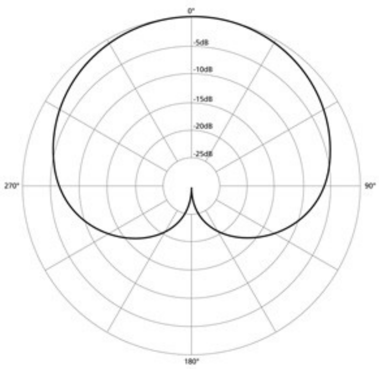
Microphones

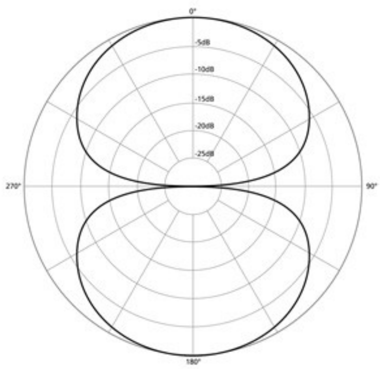
As it is very rare when there is complete silence in your home you could consider to avoid microphone use (as I do) as much as possible.

But is you need to, there a three types of microphones:

omnidirectional unidirectional bi-directional







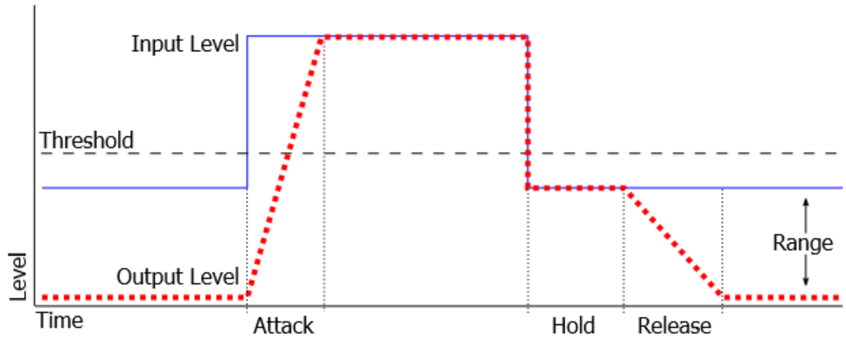
So, if you do have to use a microphone use a unidirectional one. It records sound from one direction only. This way environmental noise is reduced as much as possible.

Avoid using a noise gate

It may seem a (very) good idea to use a noise gate in your signal chain to prevent undesired hum/hiss in the audio signal.

But remember what a noise gate (or hard gate) does:

In its most simple form, a noise gate allows a signal to pass through only when it is above a set threshold: the gate is 'open'. If the signal falls below the threshold no signal is allowed to pass (or the signal is substantially attenuated): the gate is 'closed' (source: Wikipedia)



So if there is noise in the signal and you use a noise gate, this noise will not be auditable when the instrument is not played, but when playing starts, **the noise will be passed** as well.

In short, a noise gate is perfect for live playing, but you need to reconsider the use of a noise gate is recording session.

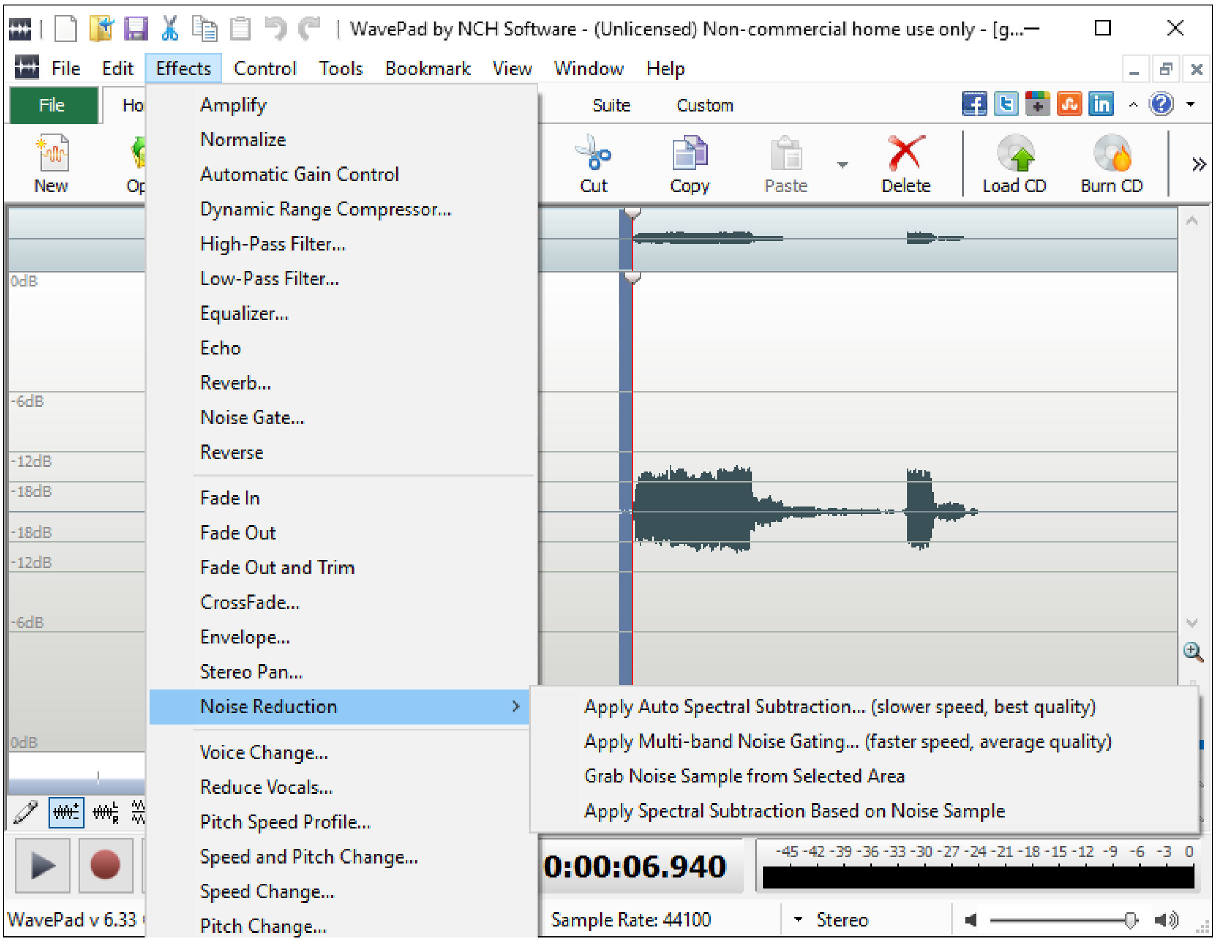
For removing noise from a recording there is an alternative method as described in the next chapter.

Removing noise from recorded audio

Once a track has been recorded and you did not use a noise gate you might notice some background noise(like hum or hiss) on the audio track. Often this is caused by either an amp or distortion and overdrive effects. In that case you may want to remove this noise from the recorded material.

To do this you need a audio editor capable of removing noise such as the free WavePad audio editor. In WavePad it works like this:

1. Open the audio file to process
2. Select an area in the audio file which contains the noise only
3. Set the amount of noise reduction (you need to experiment with this)
4. The apply the noise reduction to the whole audio file
5. Save the cleaned audio file



Using effects during recording

I use as little effects as possible during the recording phase. I prefer to add these during the mixing session because the way I have more flexibility, however, sometimes I do need to use effects during recording as these influence the way one plays an instrument. This is especially true for overdrive/distortion. Alternatively you could play with effects but record the dry signal only if you have that option. If you do not have a dry-out signal option on your amp consider the use of a splitter, where you can send the dry signal to the mixer.

Mixing

Introduction

To me mixing is all about giving each individual track its own place in the song. In a good mix the listener can identify the individual instruments. To achieve this you will need to play with volume, width and depth and placement in the stereo field.

Using group channels

This is a great way if you have multiple tracks with the same instrument. What you do is route these track to a group track and add effects, if used, in this group track rather than per individual track.

Working with MIDI tracks

If you are using midi tracks, make sure you are using VST instruments for sound reproduction because your midi tracks will sound so much better than when you use plain midi wavetable synths. There are plenty free and commercial VST instruments out there that really sound good. Some examples of free VST instruments are: Plug-ins from Maxim Digital Audio (Great keyboards such as Fender Rhodes, Grand Piano, Yamaha DX100 and Roland JX10) These are 32 bit versions but some of these are also available in 64 bit format. DSK Music (lots of instruments, only as 32 bit including bass, pads, strings etc). For synth's, strings and pads I use Syntronic from IK Multimedia and when you have Cubase 7 or higher HALion Sonic from Steinberg is included.

For drums I use Steinberg's Groove Agent). It is not only a virtual instrument with many drum kits, but is also capable of playing patterns and has a 'humanize' option, this makes midi drums tracks less sterile. I also use the electric piano module (NEO keys) from Steinberg simply because it has a very convincing Fender Rhodes sounds.
As for bass I currently use the free electric P-bass from Ample sound.

For all midi tracks you should experiment with the velocity. I have notices that many instruments sound better when the velocity is a little bit lower or higher depending on the sound you are after.

Using compression

During the mixing stage I only use a little compression on the bass and the drums. This way the rhythm section sounds tighter. For drums I use the Vintage Compressor or the standard compressor from Steinberg and for bass the TDR Feedback Compressor II from Tokyo Dawn Records. Also see the next chapter on parallel compression techniques.

Settings for drums(vintage compressor):

Settings for drums (Standard compressor):

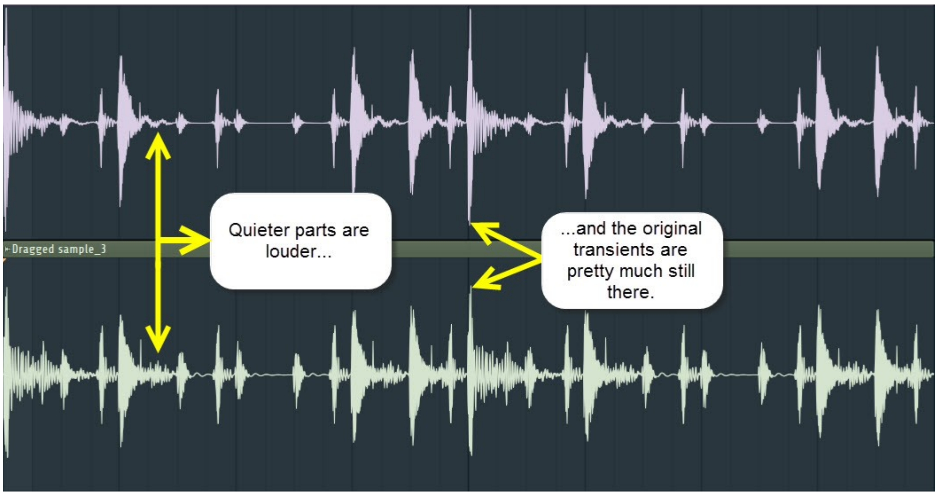
Settings for bass:

For me a threshold of about -25dB and ratio of about 3:1 to 5:1 does the trick for bass.

Parallel Compression

Parallel compression is a technique used in the mixing stage. Parallel compression is achieved by mixing(blending) an unprocessed signal with a heavily compressed version of the same signal. Parallel compression is perfect for controlling the dynamics of a sound without making it sound unnatural.

This is what happens:



This trick will add power to the audio track.

For natural sounding compression start here:

- **Ratio**: 3:1
- **Threshold**: For subtle compression you'll want 3-5 dB's of gain reduction. Watch the meter!
- **Attack Time**: Fast for thickness (2ms or less), slow for punchiness (10ms or more) When in doubt, stick with slow.
- **Release Time**: Auto or slow (above 100ms).
- **Gain**: You want it to be about as loud as it is with the compressor turned off. Bring the gain up until it's about as loud as it was before.

For aggressive compression:

- **Ratio**: Anything from 8:1 to 20:1. These are aggressive settings, so blend this compressed signal in with caution. Also, not all compressors can operate in these ratio ranges.
- **Threshold**: Aim at 5-15 dB's of gain reduction. Watch your compressor's meter for something in that range.
- **Attack Time**: Fast attack, below 2 ms.
- **Release Time**: Start with around 60 ms and tweak it until it sounds good.
- **Gain**: You want it to be about as loud as it is with the compressor turned off. Bring the gain up until it's about as loud as it was before.

Limiting

Limiting is somewhat related to compression. The main difference is that a **compressor** compresses the dynamic (volume) range. A **limiter** limits the amount of a signal passing through, but instead of taking the volume overage and compressing it. Brickwall **limiting** has a very high ratio and a very fast attack time.

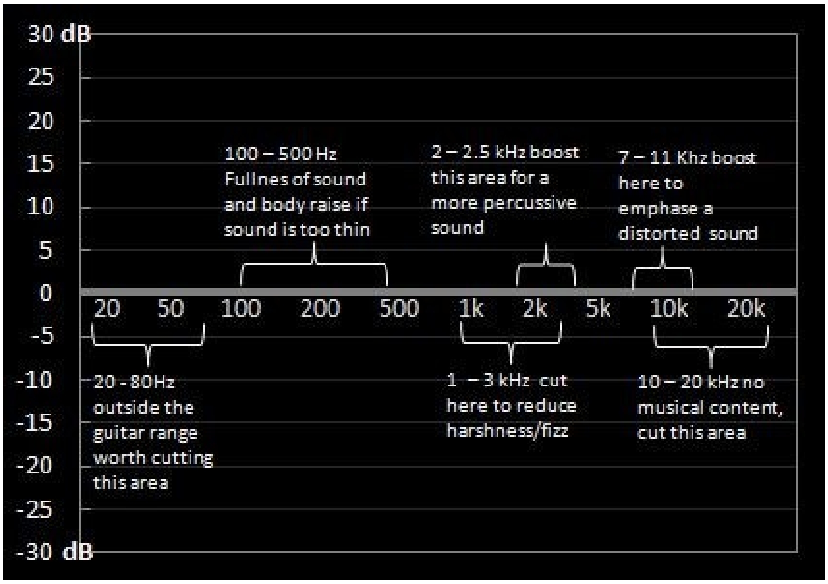
I use brickwall limiting to eliminate peaks in volume for solo instruments

Furthermore, I put the brickwall limiter as the last in the chain.

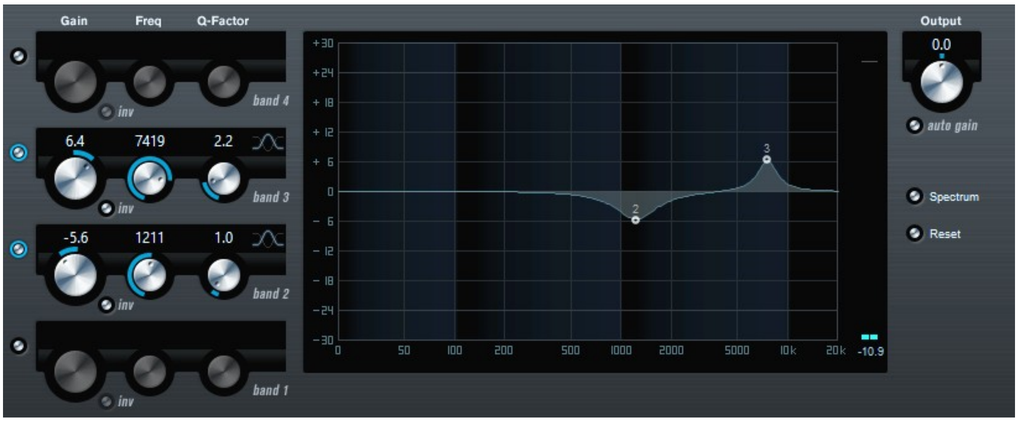
Track EQ

As a rule of thumb: Cut out the less desirable frequencies rather than boost weaker frequencies. If necessary you can raise the track volume back up to its desired level. And of course there are situations where you really want to boost, especially if we take a look at the electric guitar.

Specific EQ guidelines for the electric guitar are:



You may already have cut the low and high frequencies (as described in POD HD tone building manual). Take a look at the frequency ranges and adjust the EQ curve as needed. A typical additional EQ curve could look like this:



Here you see a cur around the 1.2 kHz area to remove some harshness and a boost at 7.4 kHz to add some brilliance to the guitar sound.

Adding clarity with EQ

The lack of clarity often occurs when different tracks are operating in the same frequency space one trick is to use inverse EQ matching. Basically it means that you analyse the frequency spectrum and create an eq curve the emphasis these frequencies on one track and cut these on the other track.

Example: Left image is Bass, Right Drums

Another way of adding clarity is to use EQ to remove annoying frequencies from a track. The way to do this is:

Create at the low end a very narrow EQ reference point with 20 dB and move this gradually along the frequency spectrum until you encounter annoying signals. (if you don't find it you're done with this track.
Narrow the Q so only the annoying frquency is selected
Set the gain to zero
Replay the section and slowly cut the selected signal until the noise gone
Beware that there may be more than one noise issue on a track

Note that room acoustics may also create undesired noise, to eliminate this use headphone to do this.

Warning: apply the previous tip only if you notice annoying frequencies of a track while listening otherwise you could end up with a Comb like EQ resulting is a flat uninspiring dull sounding track

Finally some general EQ tips

Look at the 400-500 Hz range and carfully cut a few Db's.
For basses, bring out the harmonics at 700 - 1200 Hz
Electric guitars will benefit from a cut in the 2 - 4 kHz (using above mentioned method)
Piano a small boost at 5kHz will emphasis the transients

Widening a track

Sometimes you will need to make an otherwise dull track more attractive/pleasing and widening is a good option.
The basic technologies for widening a track are:

- Mid/side editing
- Delaying one channel
- Double tracking

There are multiple free stereo enhancing VST plug-ins that can help here. Personally I like to use the Ozone Imager plug-in from Isotope

Another option is to use an automatic double tracking technique (ADT). The Proximity plugin from Tokyo Dawn Labs is an example. This plug-ins allows you to shift the frequencies of both left & right channels (among some other features)

The ADT technique was developed at Abbey Road Studios by engineers recording the Beatles in the 1960s. John Lennon was not satisfied with his own singing, he found his voice sounded too thin. To free John Lennon from having to sing everything twice for real double tracking they came up with an artificial replacement: they sent the original signal to another tape machine and re-recorded it.

Mono to Stereo

This is another form of widening a track. There are many free plug-ins to choose from such as Mono to Stereo from Steinberg (Included in Cubase). The aim of these plug-in is to make a stereo image of mono tracks and help to place the sound in the right place by the use of dedicated internal filters and pan controls.

Adding depth to a track

Where widening(see previous topic) is about left/right, adding depth is about front/back placement.

The louder an instrument in the mix, the closer it will seem.
The more reverb and/or delay, the more distant it will sound
Adding modulation effects such as chorus, flanger etc will create more distance

Using Exciters

Let's start with the definition from Wikipedia:

An exciter (also called a harmonic exciter or aural exciter) is an audio signal processing technique used to enhance a signal by dynamic equalization, phase manipulation, harmonic synthesis of (usually) high frequency signals, and through the addition of subtle harmonic distortion.

A more simple explanation: exciters were initially intended to restore high frequency loss, the demand shifted towards other tasks such as presence and definition in the (upper) mid and getting the low-end right is the key in a modern production.
In other words it is capable of enhancing the original signal in such a way it sounds better. You can use them in individual tracks as well as during mastering, and as with every effect use it in a subtle way.

There are multiple good free VST's such as the Saturation knob from Softtube

Panning

If we are happy about how our individual tracks we now need to position them(panning) in the stereo field. With panning we need to place instruments only a little bit to the left or right to create space. As low frequencies are non directional, you are better off to leave instruments as bass and kick drum centred.

Using Impulse Response to improve a guitar sound

An impulse response (IR, for short) emulates a specific cabinet speaker, including the room and used microphone(including placement. This information is stored in a file and this file can be used in Plug-in. An impulse response does NOT change the tonal characteristics of an amp but only the speaker cabinet behaviour.

Using IR's on a direct recorded guitar track will improve the sound as it adds the imperfection of a specific speaker cabinet.

For this I use the free VST plug-in from STLTones. This plug-in allows me to choose 2 different speaker cabinets (not included in the plug-in but also free available on the internet)

Stereo versus Mono Mixing

First of all: **always check the mono compatibility**. Even today recordings should be mono compatible. This way you can make sure the recording still works for mono listeners. Remember that portable radios are often mono and all car FM radios automatically switch to mono whenever the signal becomes weak.

But there is more: When you compare the stereo and mono recordings, the monophonic one loses the stereo content, but it should still have some depth and there should be no significant frequency loss. If it does there is a phase issue. In extreme cases some elements may completely disappear in mono. If this is the case to back to mixing and find/correct the phase issue.

Finally, if you can get the mix to sound good in mono first, it will definitely sound great in stereo too. If not you might need to go back to the actual mixing and review the individual tracks and look at things as compression, delay and reverb.

Relative Output Levels

The section is about the relative levels of the individual track in respect to one another. This topic is very much related to the genre of music, so use these tips only as you feel appropriate.

As a general starting point: Start mixing per individual track and take the drums first. Set the output level so that the loudest part is about -15dB. Secondly adjust the bass mixing level also to -15dB. When played back together the output will around -12dB. Now proceed with the rhythm tracks. These should reach a level of -20dB. Finally the lead instrument (and vocals). Set these to -17 dB on the stereo meters. Then open all channels and listen to the result. Mind you this is just a starting point. Keep an eye on the output meters and make sure you are not hitting the clipping point (if so lower the mixing levels) As a rule of thumb the maximum output level should not exceed -6dB, more on this in the mastering chapter

Mastering

The purpose of mastering

The purpose of mastering is to balance audio elements of a stereo mix so it will sound good on different audio systems and also that related tracks sound consistent. In order to be able to do this the stereo track you are going to master needs to have “headroom”. (More on this topic in the next chapter)

A secondary purpose of mastering is to make sure the different songs are coherent in terms of volume(loudness) and ‘character’ so that the full album can be seen as one product rather than a collection of unrelated individual songs.

General Tips

During recording and mixing I keep the individual tracks between -15dB -20dB, the mixed down stereo track will be around -6dB. So the starting point for this Mastering stage is a stereo track with a maximum level of -6dB.

As the song needs to sound good an any audio device it is useful to check this. When I am mastering I frequently check the intermediate results on various audio devices ranging from my home studio monitors to the high-end audio in my living room and my car stereo set.

Furthermore: watch out for ear fatigue. If you are happy with a song let it be for a few days, do something else and then listen again. Don’t be surprised if you are not as happy with what you hear as you were the previous time.

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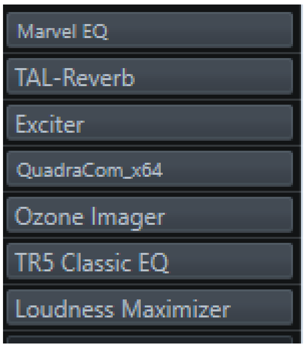
You may have a commercial recording that you like. Use this as a reference during mastering.
NOTE: Remember the earlier chapter on Monitor placement and volume, these guideline apply here as well.

Mastering Plug-ins



For mastering I use the following order of (free) plug-ins: Mid/Side Equalization, Reverb, Exiter, Stereo imaging, Multi-band compressor, Exciter, Reverb and as the last plug-in a Maximizer (compression) and, if required, dithering.

All of these components will be discussed later in this document. While recording and mixing I keep the individual track at -12dB and the mixed down stereo track will be around -6dB. So the starting point for Mastering is a stereo track with a maximum level of -6dB.

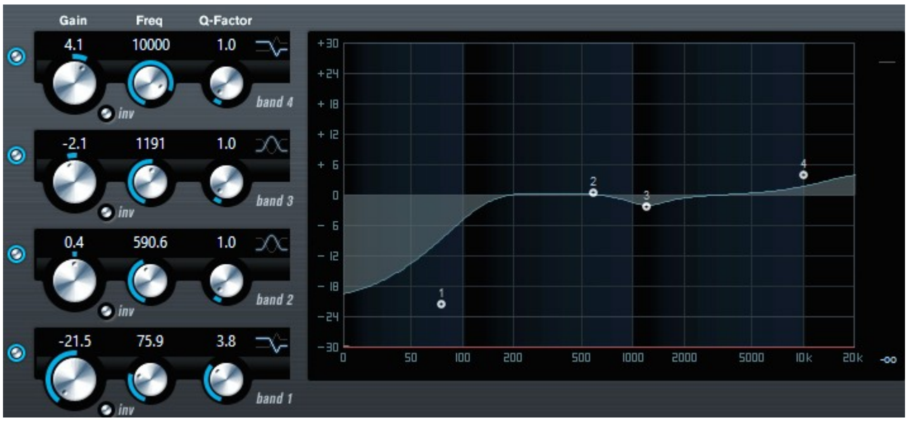
NOTE: These tools are only necessary if you need them in order to correct something, don't use it if you don't have to.

Alternatively, there are some all-in-one free mastering plug-ins worth looking into such as:

Typically these plug-ins do most of the required work and other plug-ins are often not required other than EQ and maybe a limiter at the end

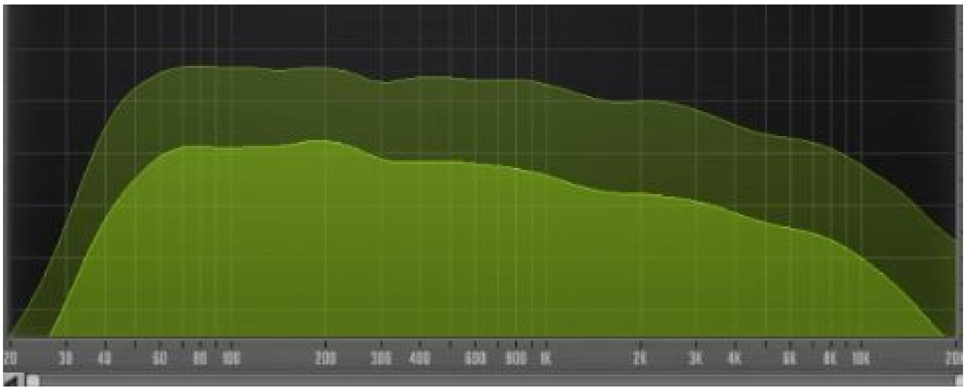
Equalization during mastering

The purpose of EQ during mastering is to prepare the track for further manipulation. In general, I roll of f the low-end, where the bass drum and bass live, because during the multi-band compression we will compensate and give this band a tighter feel. In general cutting or boosting levels (other than the mentioned low-end roll off should not exceed approximately 5dB, if you need to: go back to the mixing stage and solve the issue there.



Using a spectrum analyser

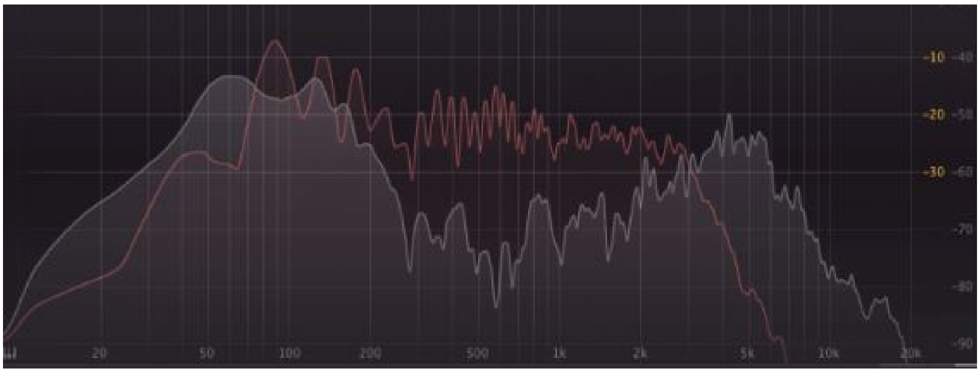
A mix that's well balanced will typically have an even distribution of energy across the frequency spectrum.



With a spectrum analyzer on your mix bus, you can quickly visualize the shape of your mix.

Look for any significant peaks or valleys. These areas might need to be addressed in your mix. This technique is particularly useful in mastering, where it can help pinpoint frequencies to cut or boost with EQ.

You can also use a spectrum analyzer to compare different tracks. (You need an analyzer with a sidechain input.) This technique makes it easy to identify exactly what tracks are different, then you know exactly how to EQ these tracks. In the end, all tracks that belong to the same project will have a similar sound spectrum.



Mid/Side processing

If you are using an EQ plug-in with mid-side processing (m/s processing) capabilities such as the free VST Marvel GEQ from Voxengo, you can do an amazing EQ-trick with this.

An explanation from Izotope:

The Mid channel is the centre of a stereo image. When the Mid channel is boosted, the listener perceives a more centred (mono) sound to the audio.
The Side channel is the edges of a stereo image. When the Side channel is boosted, the listener perceives a more spacious (wider) sound to the audio.

So basically what M/S processing allows is lowering frequencies the mid channel and boosting same frequencies in the side channel, the audio becomes wider. (or narrower if you do it the other way around). And as with all plug-ins use it gently, a few dB's is often sufficient.

Example: Suppose I want to increase the perceived volume of my Bass and Kickdrum. I go to Mid processing and add an EQ boost in the lower register (say 300Hz). To make this boost more apparent go to Side processing and add a cut in same region. This will allow the bass and kickdrum to punch through more in the mix.

In the next image (Marvel GEQ) you can see another example of mid and side EQ settings:

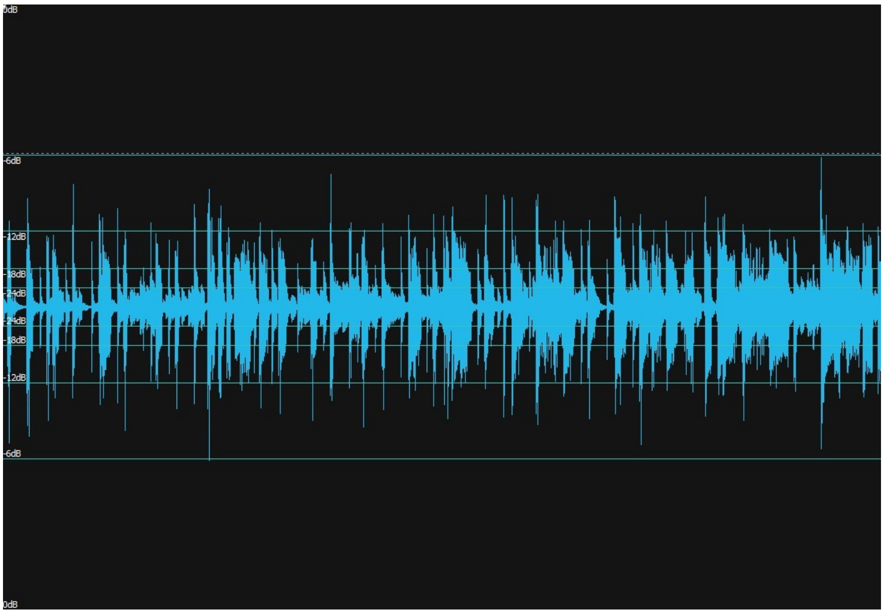
Stereo Imaging

The purpose of stereo imaging is to make the final result wider. Proximity from Tokyo Dawn Records (or Lab) has some nice options. Do experiment and as always: do not over do it.

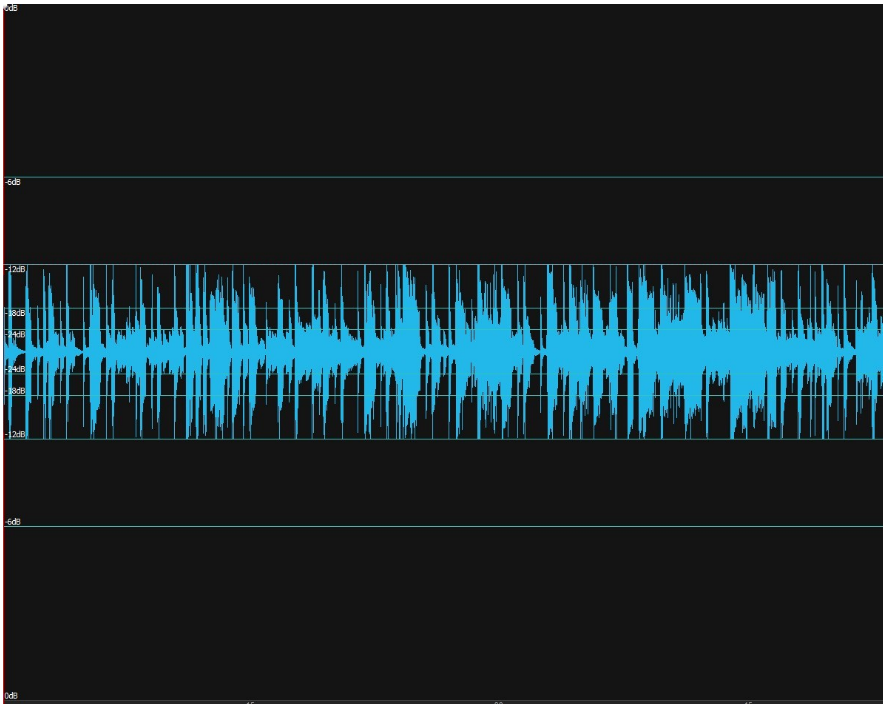
Eliminating Peaks with a brickwall limiter

If you want al loud master (as most people do) you need to look for peaks in you mix. These are often introduced during the mixing phase, especially in the drum track.

Typically a mix with peaks, introduced by drums looks like this:



Notice that the body of the mix is quite narrow, due to these peaks, With the use of a brickwall limiter you can change this into the following:

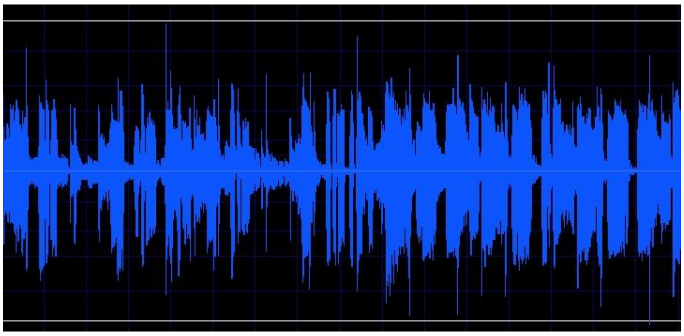


Now we have more headroom for compression, resulting in a louder mix.

(multiband)Compression

In the mastering stage of recording there are two compressors used. One is a multiband compression where we determine the required compression per band. The second one is mend to make the final track as loud as possible (or required). I keep the output level at -0.3dB to prevent clipping on cheaper audio devices and at -1dB if I master for compressed formats such as MP3.

Without mastering your mixed track will look something like this:



This track has only been normalized, which means the loudest parts are (n this case) at -0.3dB's. Notice the dynamics in this example. There is quite some difference in loud and quiet passages

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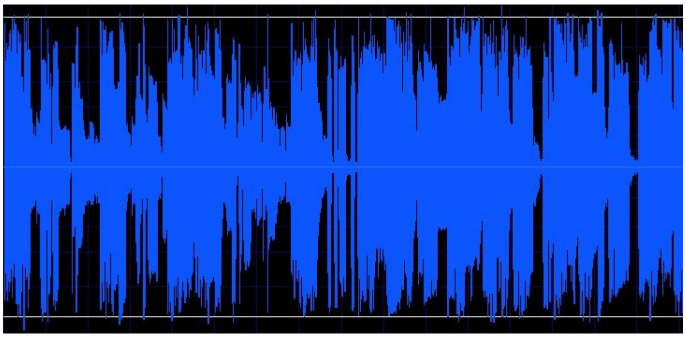
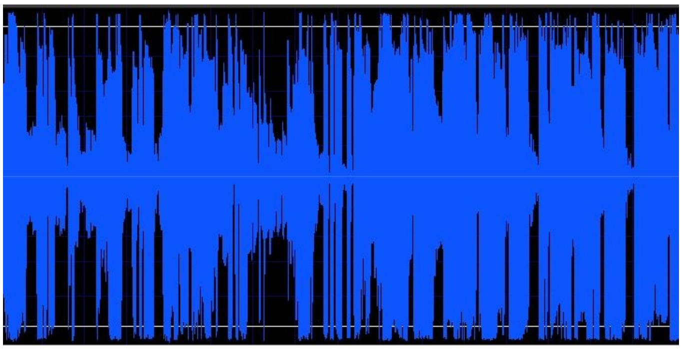
The second image is a mastered version of the same mixed track:



Notice that the dynamic range is less and as a result there are more parts that reach the 0.3dB level.

In the last example I made this track even more compressed:



In these three examples we see an increased compression rate and this will raise the perceived loudness of the track, at the same time the dynamic range is becoming less. Choosing the right amount of compression will depend on the type of music (and of course personal preferences)

Exciter

The use of exciter has been discussed earlier and the use during mastering depends on whether you need or want to add some overall harmonics or not in other words exciters can add some 'sparkle' to the mixed track

A related area is transient enhancement. A transient is a short-duration sound at the beginning of a waveform that occurs in musical sounds (source Wikipedia). A good free VST is **Bittersweet** from **Flux**.

Turning the knob to Sweet side reduces the transients which commonly decreases any transient-rich percussive sounds in the mix, and turning the knob to the Bitter side magnifies the transients which commonly increases any transient-rich percussive sounds in the mix. I use this plug-in mainly on drums and percussion

Reverb

The purpose of reverb during mastering is to glue individual tracks together. There are many beautiful free reverb plug-ins on the market. Some of my favourites:

TAL-Reverb by TAL software
Roomworks SE by Steinberg
Room041 by Analog Obsession
Raum by Native Instruments

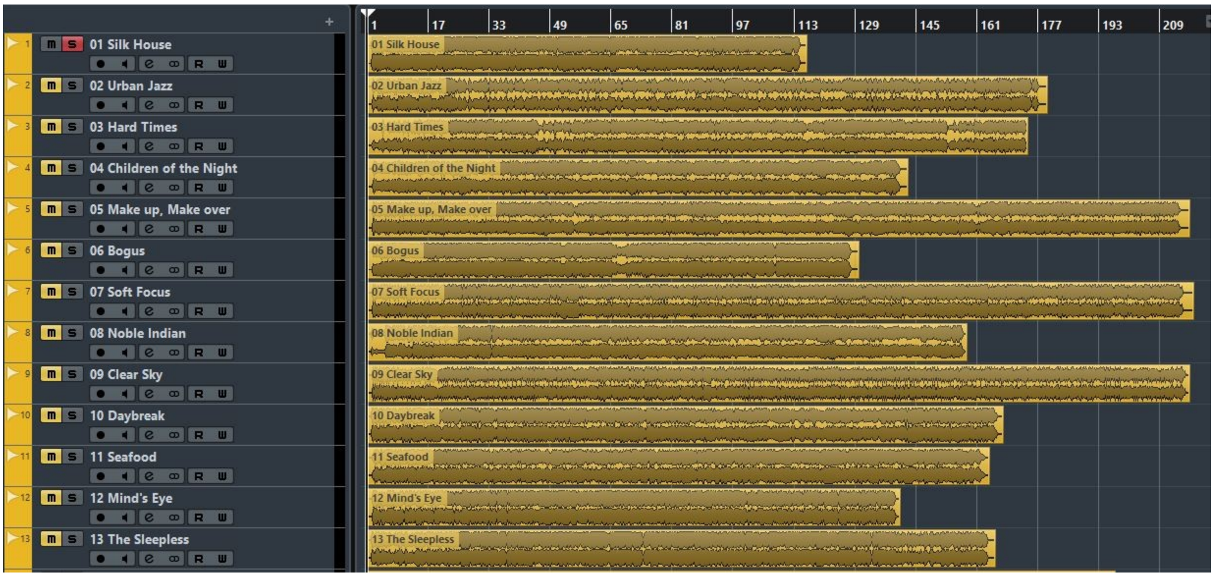
Use it with care, a tiny bit of reverb can already do wonders and, if this parameter is available, keep The following in mind: In general the pre-delay should be under 50ms otherwise the track may become blurred as the reverb itself will become a separate sound.

Another way to use pre-delay is to calculate the time with the following formula: 60.000 / BPM of the song / 8. I find this calculation not useful for drums as it tends to blur the individual drum components, For drums I use a very short pre-delay 2-5ms

Checking the consistency of the album

OK, we have mastered the individual songs, now we need to check how these songs relate to one another. This is an important step because we want the album songs to be consistent in respect to one another.

The way I do this may not be the best way but it works perfectly for me. I load all album songs in my DAW:



I mute all channels but one (in this case the first song) then I start listening to this one song. At any time I can switch to another song and thus compare the volume and feel of that song. This way I can quickly determine what songs (if any) need some further adjusting.

Dithering: The very last step

Preferably all the mastered tracks are in 24-bit audio, but in order to meet the audio CD standard they should be 16-bit. The technology behind this conversion is called dithering. I am not going into the technical details (there is enough to be found on the internet) but dithering will reduce the word length from 24 to 16 bits. If you don't do this, you will end up with unwanted distortions as a result of truncation, and although they may not be obvious to everyone, they may well become very obvious during any fade-outs or fade-ins.

Cubase comes with the UV22HR dithering plug-in, but there are multiple free dithering plug-ins available on the market.

NOTE: NOTHING SHOULD TOUCH THE AUDIO AFTERWARDS