FH MMA SALZBURG – MUSIC PRODUCTION

MIXING & MASTERING TIPS

1. MIXER – SIGNAL ROUTING TIPS

1.1 SETTING THE INPUT LEVELS

- select the proper input source (microphone / line switch);
- remember to activate phantom power for condenser microphones;
- do not activate phantom power for dynamic microphones, and especially not for ribbon microphones, or you might damage them (except for active ribbons);
- if possible, always use balanced cables to keep your signals clean and minimize undesired noises (especially ground hum, interference by induction, etc.);
- set the channel fader to 0 dB (= at Unit) and use trim/gain to set the channel input level, checking that the signal does not clip (= you get distortion) or is too quiet (= you get noise);
- you might want to press the *solo* switch with PFL (*Pre Fader Listening*) to adjust levels using the master meters on a mixer that does not have separate meters for each input;
- for line signals, start from complete left (counter-clockwise) and move right (clockwise) until the desired level is reached;
- for microphone signals, you might try to start with the knob in the middle position (12 o'clock) and then adjust left or right as required;
- remember: usually condenser mics have higher sensibility than dynamic ones;
- do not use the channel fader (= channel output) or ever worse the bus fader to adjust levels: if your input gain is too high you will get distortion even if you lower the channel and/or group fader; if your input gain is too low you will get noise even if you push up the channel and/or group fader.

1.2 PAN AND BALANCE

- beware of the difference between panorama (on mono signals) and balance (on stereo signals), and separate panorama control (on stereo signals)
- *panorama* positions a mono signal across the stereo field;
- *balance* just adjust the relative level of the L and R signal; the L channel is always panned "100% L" and the R channel is always panned "100% R";
- to adjust the stereo-width of a stereo signal, you must route it to two separate mono inputs and use the pan controls (more under);
- In Cubase/Nuendo you can use the *stereo combined panner* to achieve this (switchable in the console).

1.3 ADDING EFFECTS: *INSERT* AND *AUX SEND/RETURN*

- generally, *EQs*, *filters* and *dynamics* (*compressor*, *gate*, etc.). are used as channel insert (if stereo, also as group or master insert), to process the sound of each track / group separately;
- *delays* and *reverbs* are usually used as aux send/return, to spare processing power and/or to make the same effect available on more channels; the amount of delay and reverb can still be adjusted separately for each track with the aux send controls;
- *modulation effects* (*chorus*, *flanger*, *phaser*, etc.) and *overdrive/distortion effects* can be used in both ways; if they are used as insert, you use the *mix* parameter to adjust the balance between the *dry* (unprocessed) and *wet* (effect) signal; if you use the effect over *aux send/return*, you adjust the amount of effect with the aux send (the

effect is set to 100% wet in this case); note: chorus, flanger etc. usually sound best when the mix balance is set to about 50% dry / 50% wet;

- there are of course exceptions, where effects normally used as insert can be also used as send; sometimes you should also be creative and experiment with unusual combinations!
- for example, there is *parallel compression*: you can send all instruments of a drum kit to the same group (and even the bass), then use extreme compression settings and add some of this *extremely compressed* signal to the original uncompressed signal – it is also called *New York Compression* and can sound terrific, as the sound has all the nice transients of the unprocessed signal, and the fat punch of the compressed one;
- important: when you use any effect as aux send/return, you should always check to have the *mix* parameter set to 100% wet (effect), or else dry signal would be added to the original dry signal (potentially causing comb filtering, if the effect delay compensation is turned off)
- Most third-party plugins in DAWs like Logic and Cubase do not *know* whether they are used as insert or aux, so you will have to check the *mix* parameter yourself.

1.4 USING EXTERNAL (HARDWARE) EFFECT UNITS

- you can of course use the aux-return (as standard) to re-insert the signal from an external effect unit (for example a reverb) in the mixer, but it might be more convenient to use a free couple of mono-channel inputs, because:
- you can finely adjust the stereo width with the separate pan controls on the L and R signal;
- you can change the color of the effect sound with the EQ (works great on reverbs and delays);
- you can also send again the FX signal to another effect processor with another aux send (for example send to aux 1 = delay, and send again some of the delay return signal to aux 2 = reverb).

1.5 CONNECTING A 34-CHANNEL 8-BUS MIXER

- use the 3x 8 *bus outputs* to connect to 3x 8 Ch. multitrack device (note: you can group and submix signals, but you can record max 8 tracks at once);
- use the *24 Direct Outs* if you need to record all 24 channels at once;
- use the *Master Output* (main mix) to connect to a stereo mastering device;
- use the *Control Room out* (= Regie) for the monitors of the control room;
- use the *Studio Out* for the speakers or headphones in the recording room.

1.6 IN-LINE MIXER DETAILS

- there are two effective inputs per channel – so a 24 Ch. In-line mixer has a total of 48 inputs; this saves a lot of place, at the cost of not having full EQ features on every input;
- for example, on Mackie 8-bus: you have 24 A-channels and 24 B-channels, also called *tape return* as they are normally used to listen to the playback of the multitrack tape-machines;
- using the *flip* switch you can toggle between A and B channels, so you can for example use the main EQ on the tape return signal;
- you can also *split* the EQ between A- and B-channels, for example use the 2 peak parametric filters on the A-channels and the two shelving filters on the B-channels.

2. MIXING TIPS

2.1 WHAT IS MIXING?

- mixing is not just a technique: it is more like an art through which the musical idea of an artist/composer can be shaped into something special, that will awake emotions in the listener and make it unforgettable … at least, ideally;
- sometimes, bad mixing can completely ruin a decent recording;
- you should always keep your *musical goals* in mind; the *technical* mix parameters are your tools to achieve those goals (see under).

2.2 *TECHNICAL* MIX PARAMETERS

these are the *tools* at your disposal to *sculpt the sound* and find the right place and space for each instrument and voice in the mix:

- *levels/balance* (setting the relative levels between the musical elements)
- *panorama/width* (positioning the elements within the stereo field and setting the stereo width of stereo signals)
- *height* (position of an instrument in the frequency range)
- *colour* (adjusting the spectrum and formants, using filters and EQs)
- *dynamics* (adjusting the dynamic range, using compressors, gates, limiters, volume envelopes, etc.)
- *depth, dimension* (adjusting the forward/back position of musical elements, using ambience effects such as hall, delay, etc.).

2.3 *MUSICAL GOALS* IN A MIX

- *focus*: keep the listener's attention on the most important elements, and avoid to have “too many things happening at once”: it can be confusing and/or cause listening fatigue; of course, the number of elements to focus on and the perceived *complexity* depend on many factors, including music style, target audience and environment, etc.
- *interest*: keep the listener's interest awake, introducing a few new elements as the track develops and keeping enough variation in sound throughout the track; this is particularly important if the track was not arranged very well and potentially boring.
- *personality*: make the mix sound personal, unique and unlike anything else: try to find what is special, non-standard about this particular music piece, and shift this unusual element into focus instead of trying to hide it away; try to surprise the listener with something new; learn the “rules”, and know when it is time to break them.

2.4 PLACES FROM WHERE TO START MIXING

- Before starting mixing, you should listen a few times with all channels open to get an idea of the song/track;
- there is no “rule” from what element you should start mixing, however these are some common choices:
  - from the bass drum or snare drum (or the drums in general)
  - from the bass
  - from the lead vocals
  - from the main instrument in the arrangement
  - from a combination of the above

- if it is a typical song, the vocals should be added as soon as possible, as all other instrument will relate to the vocal track sooner or later anyway; if you try to add the vocals at the end of the mixing process, you might not have any *space* left in the mix where to fit the vocals in;
- if it is a soundtrack with orchestral sounds, you might want to start from the most important melody line (for example, the violins), or from the bass, which is the foundation supporting all the harmony;
- if it is a dance track, you almost certainly want to start with drums, then bass line and the most important synths and rhythmical elements.

2.5 SOUND OF SINGLE INSTRUMENTS AND TRACKS IN A MIX

- they do not necessarily have to *sound nice* when listened in *solo* mode: in fact, it is likely that an instrument that sounds too nice and full in solo does not fit well in a mix;
- if all elements sound too fat, too wide, etc., instead of a nice transparent mix you get a bombastic, muddy sound; but of course, if that is what you are aiming for, go for it!
- the single tracks/instruments should integrate in a complementary way with all the other elements to create a balanced, full overall sound; it is important to find their place in the mix, considering frequency range, panorama, depth, etc.
- for example: you might use a low shelving EQ at 100-200 Hz and cut as much as -6 dB from the bass frequencies of a piano track that is playing chords, to leave more space for the bass drum and the bass guitar; if listened alone, that piano will sound rather thin and not very natural, but it would work perfectly well in the mix together with the other instruments.

2.6 HOW TO PREVENT INSTRUMENTS "FIGHTING" WITH EACH OTHER

- changing the arrangement (always the best way!);
- muting one of the instruments (do not let them play at the same time);
- lowering the level of one of the two instruments;
- using very different EQ settings, emphasizing different formants (also called *frequency juggling*);
- using the pan to position the instruments in a different place of the stereo field;
- using different level of ambience (dry, or with different types and amount of reverb, delay, etc.), so that you can have some music elements more to the front and others more to the back.

2.7 POSITIONING SOUNDS IN THE STEREO FIELD

- avoid to pan mono signals hard Left or Right, it sounds very unnatural when listening with headphones and it is not necessary: when you pan something 90-95% R or L, it already sounds like it is coming from only one loudspeaker anyway …
- try to use many intermediate pan positions as well;
- try to keep the most important elements close to the center, or in some listening situation the balance between those might be completely wrong;
- typical instruments to keep middle: BD, SD, Bass, Main Vocals, Solos;
- typical instruments to have open in stereo: piano, strings, pads, background vocals, the return lines of stereo effects, etc.; use different stereo widths;
- typical instruments to position at different degrees left or right: guitars, synth lines, toms, percussion, cymbals, etc.

2.8 SETTING THE STEREO WIDTH ON STEREO SIGNALS

- do not pan all stereo signals hard L/R: there are not just mono and stereo signals, but also different degrees of stereo width;
- it is not good to just mix all keyboards and synths in stereo - you get something called *big mono*, with absolutely no panorama structure or definition and causing listening fatigue (because the brain constantly tires to localize the diffuse sound sources);
- therefore, either use two mono inputs for stereo signals, so that you can control the stereo width with the separate channel pans, or use the *stereo dual panner* (in Cubase and Nuendo), or a plugin to set the stereo width like the Waves S1 Imager.

2.9 THE MIXDOWN

- try to keep the audio resolution as high as possible throughout the signal path - so if possible record at 24-bit, mix at 32-bit float and mix down at 24-bit;
- it will be up to the master engineer to maximize the dynamics of the track, set the proper compression, and finally dither down to 16-bit for CD production;
- you might use some sum-compression, but be moderate: additional compression can always be added at the mastering stage, too much compression cannot be taken away;
- if you mix digitally, check carefully that the master out is not clipping (which can easily happen if you use a lot of tracks); on most modern DAWs, just lowering the level of the master fader will fix the problem (as most DAWs work internally at 32-bit float resolution, there cannot be internal clipping);
- best is of course to set the level of the single channels properly; as an orientation, you might try to have BD, SD and bass all hitting around -5 to -7 dB in the channel output;
- another solution might be to split the signals in different groups: you could use one stereo out for drums and percussion, one for vocals, one for all instruments and one for the effects ... and then mix everything together in the master out; in this way, you can also easily adjust the level of the most important groups in the song;
- for safety you might want to do a *vocal up* version, with the vocals about 0,8 to 1 dB louder, and a *vocal down* version, with the vocals about 0,4 to 0,5 dB quieter: so if the mastering engineer has a problem with the level of the vocal line at any point in the song, you will not need to redo the mix, he will just cut out the part he needs from the *vocal up* or *vocal down* version.

2.10 DON'T WORRY, WE'LL FIX IT IN THE MIX!

- ... probably the biggest lie in the recording industry!
- some *mistakes* happen already at the composition/arrangement/recording stage and can only be fixed if some parts are muted and/or replaced by other, compatible ones;
- for example, you might have a strings arrangement that "fights" with the vocals, which would force you to push the vocal too much up, use too much compression, etc.; the right solution in this case might be to take away the strings when the vocal line is there, and just leave them between lines ... or to replace the strings with a darker pad sound that leaves enough room for the the vocals.
- *bad sound* captured during recording cannot always be improved or corrected using EQ and other effects; if it does not sound right during recording, try to fix it right away instead of *hoping* it will be fixed later in the mix.

3. MASTERING TIPS

3.1 WHAT IS MASTERING?

- the process of optimizing the frequency and dynamic range of a recording so that it sounds best on most reproducing systems (including home stereo, hi-end systems, car stereo, ghetto-blaster, iPod, large club sound system, etc.);
- the process of preparing a recording for the final support media (for example, CD, DVD-Audio, Tape, etc.): this includes trimming the tracks to the exact length, setting fade ins and outs, adjusting pauses between the tracks, setting the relative level and balance of the single tracks, setting the track start and end markers (PQ editing), etc.
- mastering is also the last chance to fix things that went wrong during the production process: sometimes small edits and corrections might be performed at this stage, as well as "surgical" DSP processing to fix problems with sound, disturbing noises (like a 50 Hz hum), ev. distortion, clicks, etc.
- *re-mastering* usually refers to restoring and polishing an old or damaged master tape, using different techniques, such as denoising, decrackling, etc.
- mastering (like mixing) has a lot to do with music and style, and with taste as well; different musical styles often require very different approaches and the sound-aesthetics can vary considerably (just think for example of the difference in requirements between a Progressive Jazz and a Heavy Metal production, or between a Classical Orchestra and a Techno/Trance production ...).

3.2 TYPICAL MASTERING TOOLS AND EFFECTS

- *mastering EQ* to perform subtle adjustments to the freq. range (for example: Manley Massive Passive, Chandler Curve Bender, FabFilter Pro-Q2);
- *phase-linear digital EQ* (Waves Linear Phase EQ, FabFilter Pro-Q2, iZotope RX EQ, etc.);
- *master bus compressor* to gently control the dynamics (Vertigo VSC2, Manley Vari-Mu, Millennia TLC-2, etc.)
- *multi-band compressor* (UAD MultiBand) or *multi-band dynamic filter* (FabFilter MB) to fix problems in the mix; can also be used to achieve ridiculously high (and mostly unnecessary) loudness levels;
- *virtual tape simulation* to add subtle *saturation* and *analog warmth* to the sound (SlateDigital VTM, UAD ATR-102)
- *digital brickwall limiter* to avoid clipping and maximize loudness (SlateDigital FG-X, UAD Precision Limiter)
- *stereo imager* (Waves S1) to adjust the stereo width;
- bass/treble *enhancer/exciter* to "refresh" a dull sounding recording (SPL Vitalizer);
- *DAW* (Digital Audio Workstation) with excellent AD/DA converters;
- *audio mastering software* supporting ISRC, UPC/MCN, CD-Text, different types of fades, etc. (for example: Presonus Studio One, Steinberg Wavelab, etc.)
- very often *a combination of analogue and digital processing* is used – just choose what sounds best for the track!
- *excellent studio monitors*, possibly specifically made for mastering purposes (for example, Neumann, Genelec, Quested, B&W, Dynaudio, etc.)
- *a mastering room with very neutral and natural acoustic characteristics* (uncolored, linear frequency response, constant reverb time around 200 ms); if this is not available, resort to nearfield monitoring (so the room influence is minimized) and high-end headphones to check the low end;
- *a pair of good ears, experience and ... good taste!*

3.3 WHAT CAN (AND SHOULD) BE FIXED/ADJUSTED

- setting the track start: check that there is no unwanted pause (or noises) before the music begins, but avoid trimming the track so tight that ev. breath, air or whatever is there before the sound starts is cut away! Typically, the music waveform should start 50 to 500 ms after the beginning of the sample (shorter for pop/rock and longer for classic tracks);
- setting the track end: check that the track end is not cut too early (especially if there is some ambience or delay at the end) and use a nice fade-out even if it is only for the background noise;
- on many pop/rock tracks there is a longer fade out on the chorus: make sure to perform this as the artist/band/producer desires to have it; it is always better to do this at the mastering stage, and not as mixdown (else you might get digital noise at the end of the fade out);
- types of fade ins and outs: exp. and log. curves sound more *musical* than linear ones; use inverted *S-like curves* for longer fade outs;
- adjusting the track volume and L-R balance, also in relation to the other tracks: check that the L and R channel have ab out the same peak and RMS levels, and that the track has the correct loudness in relation to the others;
- you should never normalize every track on the CD! doing this, a *quiet* performed track would be boosted much more in level than a *loud* performed one, and the *quiet* track would end up being perceived louder/closer than the *loud* performed one.
- adjust subtle differences in balance between the different frequency ranges with a mastering EQ (see EQ tips);
- adjust the stereo width if the track sounds too *narrow* or too *wide*, using a stereo imager (like Waves S1) or a psychoacoustic processor (like the SPL Vitalizer)
- to refresh a dull recording, if nothing else works: use (with moderation) a bass/treble enhancer or exciter (but try with the EQ first);
- to remove undesired tape hiss: try a denoiser, possibly with *fingerprint* function to identify the exact spectrum of the noise to be removed; check carefully that you do not cut important high frequency parts of the signal (better noisy than dull);
- adjust the dynamic range if it does not fit the final medium and/or the final listening environment (so that you might end up having to regulate the volume all the time as some parts are too quiet, and some are too loud);
- compare the overall sound also with other productions (your *reference CDs*) to make sure you are within the range of possible variations.

3.4 MASTERING EQ TIPS

- adjusting the low freq. end: too little of it and the recording sounds *thin*, *powerless*; too much and it sound *boomy* and *distorted* on most loudspeakers; you can use a mild low shelving filter for this purpose, or a wide peak param. EQ centered around 50 Hz;
- adjusting the high freq. end: too little of it and the recording sounds dull, unclear, especially when listening at low volume; too much and it sounds very harsh and unpleasant, especially at high volume levels; you can use a mild high shelving filter for this purpose, or a wide peak param. EQ centered around 16 kHz;
- check the balance between the bass frequencies and the high ones; typically, in pop/rock/electronica the difference between the loudest bass frequencies (around 50 to 100 Hz) and the high ones (around 10 kHz) is 30 to 40 dB; if the spectrum analyzer shows less than 30 dB difference, the track might sound rather harsh, aggressive and thin; if it shows more than 40 dB difference, it is possible the track might sound very warm, but also borderline muddy and dull.
- if you need some *more power*, try boosting 16 to 60 Hz, but check this out with a system that does respond down to 16 Hz, or with good headphones, or you might overdo it! Remember that energy in this range can “eat up” a good deal of the whole dynamic available;

- if you need some more BD *punch*, try boosting 50-60 Hz;
- if the bass is too loud, try cutting 100 to 150 Hz;
- if the sound is *muddy* (especially obvious when listening on small multimedia speakers or a ghetto blaster), try to cut around 200-250 Hz;
- if the overall sound lacks some *warmth* and *fullness*, try boosting 250-400 Hz, or cut this range if the overall sound is muddy;
- if the vocals are not getting through the mix, you might try to enhance the range were the vocals have the most important overtones (800 Hz to 1,5 kHz);
- if the guitars are too *sharp*, you might reduce a bit the range between 2,5 and 4 kHz;
- if the mix is *unclear* (= not transparent), you could try to boost around 2-3 kHz; beware, 2 to 4 kHz is the range we are most sensible to: if this is boosted too much, at high listening levels it can cause listening fatigue or even hearing damage!
- if the vocals lack *presence*, you might boost a little around 5 kHz (but take care of the "S");
- if the "S" and "T" are *too sharp*, you might cut around 6-7 kHz or use a de-esser; (but it is better to use a de-esser when mixing, and better yet to position the microphone not directly in front of the singer when recording);
- if the cymbals and hi-hat sound *harsh* and *metallic*, you might cut 10 kHz and boost a little over 12-15 kHz; it sounds more elegant;
- if the recording lacks some *spark* and *finish*, you might add some *air* with a shelving at 16 kHz;
- in some case, it might be best to use *phase linear EQs*: they will not modify the shape of low-frequency waveform (that would in turn reduce the available headroom and/or cause clipping); but careful: in some situations, phase linear EQs can produce a disturbing *pre-ringing* (it is like hearing a percussive sound quickly fade in before the hit)

3.5 MASTERING COMPRESSOR TIPS

- if the overall dynamic range is too wide (too much difference between most quiet and most loud passages), you can use a compressor at moderate compression ratio (1:1.2 to 1:1.6), relatively low threshold (-20 or even lower) and *soft knee*, to *adapt* the overall dynamic range of the recording to the final audio support (for example CD);
- if just the signal peaks are too loud, you can try a compressor with high ratio (1:2 to 1:4), high threshold (-10 or higher), hard-knee and fast attack/release times, to just control those peaks; in this case, you might also want to try a limiter instead;
- sometimes you just want to adjust the level of different song parts (like the intro): in this case use a volume curve instead of the compressor;
- generally: classic and jazz productions use very little mastering compression (especially if there was already compression on the single channels and groups during mixing); for pop, rock, dance etc. you might need some additional sum compression, but beware that if you overdo it, it will sound terrible when broadcasted over the radio (where huge amounts of compression might be added) as it will have no dynamic and punch left!
- if you have a good sounding analogue compressor and decent AD/DA., don't be afraid to go out and in of your DAW and use that instead of your favorite plugin.

3.6 MULTI-BAND COMPRESSOR AND DYNAMIC EQ

- multi-band compressors should generally be avoided, as they split the audio signal in several separate audio bands; the high and low cut filter used cause changes in the signal phase; when the bands are recombined together, phase boosts or cancellations might occur in the crossover ranges (the signal loses transparency).
- multi-band dynamic equalizers, like the FabFilter MultiBand or the TDR Nova (free plugin), are definitely to be preferred: these tools cause almost no phase distortion and hardly any loss of transparency in the signal;

- multi-band dynamic EQs can be used to optimize the dynamics of different frequency ranges in the spectrum; if used correctly, it can let you reach a higher perceived volume without distortion;
- they also be used instead of a traditional EQ to balance lows, mids and highs in a track.

3.7 BRICKWALL LIMITER

- the *brickwall limiter* should be the last effect in your master bus chain, to control short peaks and avoid any clipping of the signal;
- typically, you will set the max level as – 0.1 or - 0.2 dB FS, as some older DA converters can cause distortion when the signal reaches 0 dB FS;
- up to 2-3 dB limiting can be almost undetectable and help you gain some additional loudness, but make sure there is no distortion on the loudest passages;
- set the release time as short as possible to avoid hearable *pumping* artifacts, but beware that very short release times can also cause undesired distortion;
- clipping on the master out should generally be avoided: do not try to boost the volume of your production to be the loudest of all, but go for the *best sound quality* you can get, at the loudest level you can reach *without* distortion/clipping;
- clipping on single tracks during mixing (like on drums) while keeping other tracks like the vocals *clean* is however an option, if you are looking for a certain aggressive sound on the drums (like in hip-hop, techno, etc.); so you can have single tracks clipping, even the drum/percussion group clipping, but avoid master out clipping;
- in certain cases, moderate clipping is also used instead of limiting when mastering through an analogue effect chain; this might sound ok on high-end AD converters (for example by Metric Halo, Lynx, etc.), and rather ugly on cheap ones.

3.8 NO PROBLEM, THE MASTERING ENGINEER WILL FIX THAT!

- … probably the second greatest lie in the record industry!
- of course, like in mixing, there are some things that *cannot* be fixed;
- for example, if you have two instruments in the same freq. range and one is too loud or too quiet, there is very little you can do at this point;
  example: the bass is too loud, and covers the bass drum – nothing can bring this bass drum back to life and make it punchy ... however, if the bass drum is too loud and the bass is too quiet, you could try to compress the bass drum and bring the bass (low level signal) up;
- if the mixdown was already distorted or clipped, it is almost impossible to fix this later: in this case, re-do the mixdown!

RECOMMENDED LITERATURE

- PIEPER, Frank (1999): *Das Effekte Praxisbuch* – CG Carstensen (ISBN 3-910098-16-9)
- OWSINSKI, Bobby (1999): *The Mixing Engineer's Handbook* – Artist Pro, Mix Books (ISBN 0-872887-23-5)
- OWSINSKI, Bobby (2000): *The Mastering Engineer's Handbook* – Artist Pro, Mix Books (ISBN 0-872887-41-3)
- HENLE, Hubert: *das Tonstudio Handbuch* – GC Carstensen (ISBN 3-910098-19-3)

WEBSITE

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