Zeeland Music Production

Music Production Workflow

A guide to Recording, Editing, Mixing, and Mastering

Contents

Introduction

The objective of this document is to discuss a workflow for music production, covering recording, editing, and mixing, and to a lesser extent mastering. The intended audience are music producers who see benefit from a guide on production fundamentals. The methods and tools laid out here are by no means the only or even the best way to approach things, but they can serve as framework.

Music Production Overview

As result of the transition of music production equipment from hardware to software and from analog to digital, the industry has transformed to a model that enables any person with access to computing devices to be a music producer and to be creative with a relatively minimal investment.

At the same time that transition has created a myriad of companies, brands, tools, and methods for addressing every aspect of the creation process, and an equivalent number of YouTube channels and videos providing guidance on what to use and how to use it - which can be overwhelming. This guide is loosely based on the curriculum from a renowned recording studio and educational facility.

Basic components and terminology

Music production components

- **DAW (Digital Audio Workstation)**: A software program typically on Mac or PC that facilitates recording audio signals and provides functionality to organize and manipulate the recorded audio signals and to create music with "virtual" instruments. This is the digital version of a console, which was and continues to be used in recording studios for that purpose. Examples include Pro Tools, Logic Pro, Ableton Live, Reaper, Cubase, and Cakewalk by Bandlab. This document uses Pro Tools terminology and concepts for describing DAW functionality.
- **Audio Interface:** A device that facilitates transmitting audio signals between sources (i.e., microphones, instruments), computer, and listening devices (i.e., speakers, headphones) and converts signals from analog to digital and back along the way.
- **Plug-Ins or Effects**: The analogue world uses physical devices to manipulate audio signals. These devices are often called outboard gear and have to be "plugged in" to a console. They enable adding effects (i.e., Compression, EQ, Reverb, Delay) to the audio signal. In the digital world software applications representing these physical devices and effects are referred to as plug-ins. DAWs typically include a set of "stock" plug-ins, in additional plug-ins can be purchased from 3rd party providers and integrated into a DAW.
- **Files**: The content recorded and/or created in this environment is stored as files. This includes complete recordings, e.g., a mix or master, and individual tracks within a recording, e.g., a vocal or drum track.
    - Individual files created/exported from a DAW are often referred to as "stems", the sum of individual files that make up a song are often referred to as "multi-tracks".
    - Files are stored in different formats, including WAV (Waveform Audio File), AIFF (Audio Interchange File Format), and MP3. The latter is a compressed format that uses an algorithm to reduce file size, while the former are uncompressed, meaning they contain all recorded or programmed information. MP3 was essential when data storage was expensive and internet bandwidth was low, however, is less relevant today.

Digital Audio Workstation (DAW)

A bit more detail on DAWs as are they are the center piece and main user interface for music production. DAWs were initially designed to replicate studio consoles and therefore their structure and concepts often resemble consoles, yet without the physical constraints. DAWs can interact with studio consoles and replace tape machines as storage devices, or they can completely replace a console. Over time, DAWs have evolved towards a more computer-centric "in-the-box" creation process. Structure and features can vastly differ; however, the concepts below are fairly common across DAWs:

- **Audio and Instrument tracks** contain recorded audio signals, or MIDI programming for virtual instruments.
  - Can be mono or stereo
  - Have an input and output, which links to the audio interface used with the DAW
  - Have volume and panning, mute, and solo
  - Allow inserts of plug-ins to add effects to an audio signal
  - Allow sends, which create copies of the audio signal
- **Auxiliary tracks** are used for grouping or routing purposes. They behave similar to audio tracks, however, do not store a signal. They allow inserts and sends.
- **Inserts** is a section on a track to add effects and instruments, typically from a drop-down list.
- **Sends** is a section on a track to send copies of a signal stored in the track or routed to the track to another track via a bus.
- **Busses** transport signals. A signal is sent from an audio, instrument, or aux track to one or multiple busses and is received from a bus by one or multiple audio, instrument, or aux tracks.

In the illustration below a drum kit is recorded to 5 audio tracks (3 mono and 2 stereo) and then routed to an aux track that groups all drum tracks. An electric guitar is recorded to a mono audio track. The electric guitar track and the group track for the drums are both sent to the mix bus which is then picked up by an aux track combining signals of all recorded tracks in the mix.
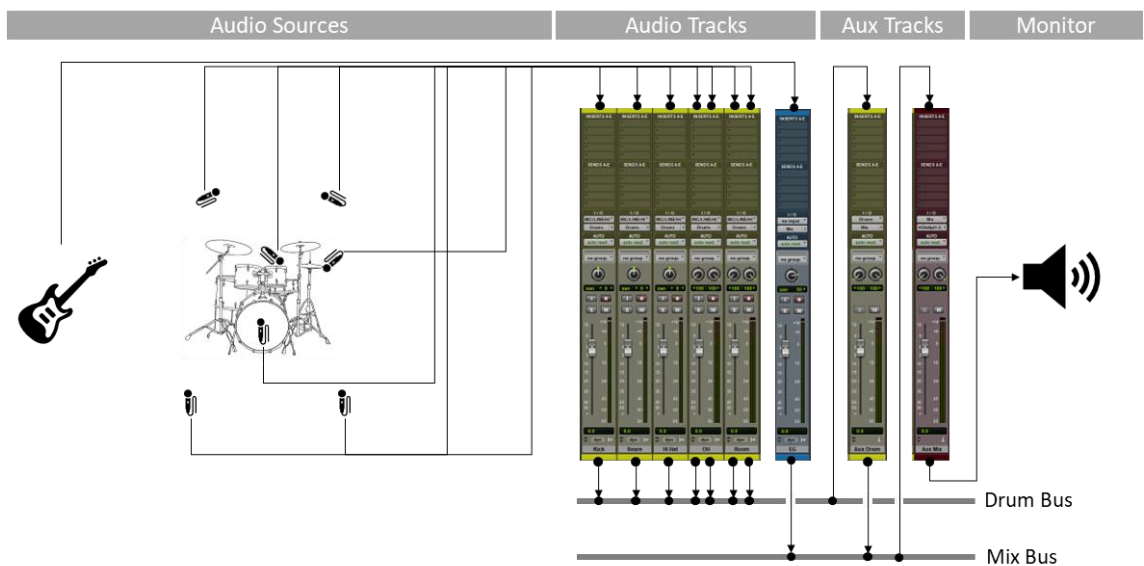
(Note that terminology might be confusing as the term Mix Bus (or 2-Bus or Master Bus) often refers to the auxiliary track that contains the mix rather than the bus itself)

Music Production Overview

Templates

DAWs will typically allow using templates for sessions or projects, which enable a user to create a structure for recording or mixing sessions once and re-use that structure for subsequent sessions. Templates improve efficiency and avoid mistakes.

Sample Rate and Bit Depth

DAWs will typically allow setting a sample rate and bit depth for a session or project. Without going into much detail: a sample rate of 44.1 kHz for capturing audio and a 16-bit depth for storing audio are sufficient for CD-quality. However, with streaming services now offering higher resolution audio, it is advisable to use 24-bit depth and 48 kHz or 96 kHz sample rates, which also overcome some technical limitations of lower sample rates. The trade-offs are file size and impact on processing. When storing audio, un-compressed formats such as WAV or AIFF will retain all audio information, compressed formats such as MP3 will use algorithms to remove information and reduce file size.

References

References are useful throughout the production process to compare a mix or elements of a mix against professional recordings and mixes, and to have a target to aim for to understand and match the characteristics of an instrument, a genre, or to get an idea what the artist one is recording or mixing for wants the end product to sound like.

Loudness

Loudness of audio signals is represented in different ways along the signal path. In the example below, the signal path for recording an acoustic guitar through a console and storing the recorded signal in a computer will transform the signal from sine wave to voltage to digital, and vice versa on the path from computer to speaker to the listener of the recording,

It is important to keep an eye on loudness of audio signals and the overall mix during throughout the process of recording and mixing, and it is one of the deliverables of mastering. The human ear perceives louder as better, which resulted in the loudness wars that raged until streaming services normalized loudness and eliminated the discussion. There are defined loudness standards for media, e.g.

Platform	Peak	Loudness	Dynamic Range
Streaming	-1.0 dB$_{TP}$	-13 to -15 LUFS	> 9
CD	-0.1 dB$_{TP}$	-9 LUFS	> 9

The goal for mastering is to meet the above standards, and the goal for mixing is to leave room for mastering to do its job, meaning loudness should be -18 LUFS or less, and peaks should be -3dB$_{TP}$ or less.

Glossary

- **dB$_{FS}$** (Full Scale): Used for amplitude levels in DAWs with 0 dB$_{FS}$ as maximum. Refers to the amplitude of a signal in relation to the maximum a device can handle before clipping occurs.

- o **dB_{TP}** (True Peak): The scale is dB_{FS} but measured with a true peak meter. dB_{TP} refers to the peak amplitude of a signal compared with the maximum a device can handle before clipping occurs. In digital systems, 0 dB_{TP} would equal the highest level the processor can represent. Values are always negative or zero since they are less than or equal to full scale.
- o **LUFS** (Loudness Units Full Scale): A standard for normalization of audio levels. It has no absolute reference and describes level differences.
- o **True Peak**: A maximum absolute level of the signal waveform.
- o **RMS**: Root Mean Square. The average power of the audio signal, resembles what human ears perceive as the loudness of audio
- o **Loudness range**: Dynamics or difference between the average soft and average loud parts. Similar to RMS but based on the LUFS scale which is optimized for human perception.

Metering

Metering is used during recording, mixing, and mastering to ensure appropriate volume levels and to avoid clipping, in other words signals reaching volumes above the sample rate limits. Meters display the spectrum between a noise floor (or self-noise) and a Maximum Operating Level (MOL) defined for audio devices. The term Standard Operating Level (SOL) refers to a target audio signals should be amplified to when recording and is +4 dB_U for professional audio. DAWs allow to choose different type of meters to monitor volume, as do compressors and limiters:

- Volume Unit (VU) Meter
  - o Non-linear scale from -20 VU to + 3 VU, with a red section starting at 0 VU, where 0 VU is the SOL, but + 3VU is below the MOL.
  - o Typically has a clip indicator since the scale does not go to clipping level
  - o VU meters are slow and need 300ms integration time to react to a signal
- Peak Program Meter (PPM) operate similarly as VUs but with faster integration times, and measure peaks vs. averages. PPMs cannot however meter true (inter-sample) due to integration time.
- Loudness Unit (LU) Meter
  - o Measure momentary (average over past 0.4 sec), short-term (average over past 3 seconds) and integrated loudness (average for the complete track).
  - o Also measure true (or inter-sample) peak
  - o LUFS articulate standards for streaming (-13 to -15 LUFS), and broadcast (21 LUFS).

During recording use gain staging within a pre-amp to record at SOL. As a rule of thumb, SOL (+ 4 dB_u) in the analog world equates to -18 dB_{fs} in the digital world. During mixing it is best to use LUFS meters (e.g., Izotope Insight) and aim for integrated LUFS of -15 dB_{FS} or lower and True Peak lower than -1 dB to leave headroom for mastering. During mastering aim for the levels defined for the target media.

Step	Target Loudness
Recording	- Analog: + 4 dB_u - Digital: -18 dB_{fs} - VU Meter: 0 VU
Mixing	Digital: -18 to -15 LUFS (integrated), with peak lower than -1 dB_{TP}
Mastering	- CD: - 9 LUFS (integrated) and -0.1 dB_{TP} - Streaming: -14 LUFS and -1.0 dBTP

If SOL cannot be reached when recording or if the noise level is too high at SOL, then consider changing microphones or microphone placements.

Music Production Steps Overview

Recording

All aspects of creating and capturing an audio signal to be used in a music production, includes

- recording acoustic instruments (e.g., strings, guitars, brass) and voices with microphones,
- recording "electric" instrument (e.g., guitars, keyboards) line signals and/or amplified signals with microphones, and
- programming audio with "virtual" instruments within a computer.

Editing

Manipulating audio signals e.g., to align signals to a tempo or to each other, and/or to eliminate unwanted noise from a signal.

Mixing

Arranging multiple individual audio signals and tracks to create a stereo or surround "mix", a process which includes

- volume leveling across tracks included in a production,
- panning signals in the stereo or surround spectrum, and
- optimizing the individual signals and the overall mix to a desired outcome.

Mastering

Preparing the audio mix for the media it will be published on, e.g., CD, vinyl, or streaming. Includes matching required levels for each medium or service, and ensuring consistency of across e.g., all tracks of an album to be released. While originally a mostly technical process, mastering is nowadays is also considered part of the creative process.

Organization

Especially for artists producing and mixing their own material it is worth considering keeping recording and editing/mixing steps separate, as in having a distinct project for recording/producing and only start editing and mixing once recording/producing is completed and using separate DAW projects/sessions for recording/producing and for editing/mixing. Mastering should then also be a separate step once a mix is finalized.

PLAP Resources
- [10 Home studio production, mixing and mastering tips](#)

Music Production Steps Detail

Recording

Recording includes capturing sound from acoustic instruments, vocals, and electrical instruments as well as programming virtual instruments "in the box". When recording audio, ensure signals are recorded at the right level (see metering) and consider alternatives for microphones and placement to achieve

optimal results based on the intended role of the instrument(s). A good recording is key to a good mix, and mixing cannot fix bad recordings.

Microphone types

Microphones transform acoustic energy into electrical energy (a voltage).

- Electromagnetic microphones: Also referred to as dynamic microphones. These microphones do not require power to operate.
  - Moving coil microphones have a sturdy build and are good for using on stage in a live recording situation. They can handle loud sources but will create a low output and are less sensitive than other microphones especially to higher frequencies.
  - Ribbon microphones are less sturdy, more sensitive, and better with high frequencies but cannot handle high volumes. Their tone is typically very balanced. Their polar pattern is typically Figure-Of-Eight (see polar patterns below)
- Electrostatic microphones: Also referred to as condenser or capacitor microphones. Require power to operate (called phantom power), which is provided by pre-amps that are either standalone devices or integrated in consoles and audio interfaces. Tube microphones also require power and typically come with their own power supply. These microphones are more sensitive than dynamic microphones, better with higher frequencies but can be bright, and are less tolerant to loud volumes.

Polar patterns

Polar patterns describe a microphone's sensitivity to off-axis signals, meaning signals that do not originate directly in front of the microphone:

- Omni (-directional): signals are picked up equally from all directions
- Figure-of-Eight (Fo8): equal pick-up on the front and the back of the mic, but not on sides.
- Cardioid: front, combination of Omni and Fo8
- Super Cardioid: wider, more Omni than Fo8
- Hyper Cardioid: deeper, more Fo8 than Omni

Proximity effect

Refers to a bass boost when a sound source is closer than 10-15 cm (4-6 in) from the microphone. Omni microphones have no proximity effect, Fo8 have the most, and cardioid are in between.

Microphone techniques

- Mono positions will be close, natural, or ambient (room) dependent on distance from the source, in other words the ratio between direct and indirect (room) sound.
- Stereo positions are spaced pair, coincident, and near coincident and allow to manipulate width, and depth, the specifics are dependent on distance and angle to the source.
  - Coincident (X/Y or M/S) provides a true sense of angle and stereo image, but not a lot of depth
    - Mics are placed as close as possible to each other but not touching.
    - X/Y place small diaphragm condensers at a 90-degree angle to eliminate phase differences. Pan hard L/R in the mix or vary panning.

- A "Bluemlein" array is an X/Y with Fo8 microphones. Pan hard L/R and consider the proximity effect. With favorable room acoustics yields an open sound and picks up more room sound than X/Y with cardioids.
- M/S (Mid/Side) consists of one cardioid pointed at the source, and one Fo8 angled 90° to cover the sides. Pan cardioid to the middle, pan the plus side of the Fo8 L, create a copy, phase invert, and pan R – levels need to be the same L and R to cancel the common signal between the two mics. The louder the L/R the wider the sound.
- Non-coincident (spaced pair A/B) will provide depth but not a strong center image
  - Place 2 mics (of any type) perpendicular to a central source at 1-5m distance to each other
- Near coincident is a combination between spaced pair and coincidence
  - Always pan hard L/R
  - There are several standards originating from broadcasting (ORTF, DIN, NOS, RAI, …)
- Some examples for mic placement
  - Vocals: Omni is the most natural and spacious, cardioid is more direct and up-front
  - Piano: using two Omni left and right will yield a natural and spacious sound
  - Acoustic guitar: M/S will give depth to an acoustic guitar that is center in the mix

PLAP Resources
- General Recording
  - [Recording lessons page](#)
  - [5 Essential recording tips](#)
- Drum Recording
  - [Miking up a drum kit](#)
  - [Recording and producing drums](#)
- Bass Guitar
  - [Home studio bass recording](#)
  - [Recording and producing bass](#)
- Vocals
  - [6 Vocal recording tricks](#)
  - [Recording vocals cheat sheet](#)

Editing
Working in a DAW, the editing process starts with loading and organizing the tracks to be edited and is the initial task before mixing. It can include the following steps

- Removing silence and noise from individual tracks.
- Aligning tracks to a tempo (the BPM of the song), also referred to as quantizing.
- Cleansing and editing vocals and background vocals, including comping, creating doubles, removing unwanted noise, aligning to tempo and beat

When editing tracks, it is advisable to create copies for every new step, in order to be able to go back to the previous steps, and to the original recordings in case of issues.

Removing silence and noise

It is common to delete silence on tracks and to apply fades at the beginning and end of parts of a track than contain a signal. Fades can be used in a technical fashion, and also in creative fashion to support the flow of the music.

Aligning tracks to a tempo and beat

This is as much a technical as a creative decision as well. For drums and other transient-heavy instruments DAWs typically have functionality to aid with aligning tracks or groups of tracks to a beat and tempo. DAWs also have elastic audio or warp capabilities to stretch or shrink audio signals without altering pitch.

Vocal editing

Vocal editing can be more or less elaborate depending on the genre and quality of the vocalist(s) and the vocal recording(s). Here are a few steps to consider, ideally in the sequence laid out here:

- Comping: assuming a vocalist will have recorded multiple takes of a track, comping is the activity to assemble a lead vocal track from the best quality words and phrases of the recorded takes. Applies as well for creating doubles of the lead vocal and, if deemed necessary, for background vocals. DAWs provide functionality to assist with comping.
- Cleansing: removing room noise, crackle, clicks, and mouth noises from all vocal tracks as well as de-essing vocals. The industry standard toolset for this is Izotope RX which offers an elaborate set of capabilities for this activity.
- Editing: removing silence, removing breath from all but the lead vocal track. Aligning vocal tracks using fades and/or using elastic audio/warp.

Vocal editing can be elaborate and time-consuming but worthwhile given the prominent role of vocals in a recording or performance.

Mixing

General thoughts

A lot depends on context, i.e., whether you are mixing your own productions or whether you are mixing someone else's production, and what the environment for mixing is, e.g., headphones, home studio, or professional studio. A few rules of thumb:

- Use references to compare your mix with a song you know well and that represents the sound you, or your customer are looking for.
- Check mixes in multiple settings, such as car, phone, studio, etc. Ask others to provide feedback.
- Check mixes in mono: in the real world a lot of listening happens in environments that are far from ideal stereo. A mix should sound good in those environments.
- If mixing on headphones, consider using Sonarworks SoundID to calibrate your headphones. If you want to get one step further, consider SoundID to calibrate room and speakers as well.

Workflow

Start by creating a basic mix, which will then be further refined towards a final mix. Templates are useful to maintain a structure than can be re-used across mixing projects.

Use auxiliary tracks and busses

- to group tracks as needed to apply effects to groups of tracks instead of or in addition to applying effects to individual tracks,
- for parallel processing (see compression), and
- to summarize all tracks and groups into a stereo mix (mix bus).

Consider using a stereo audio track as a print track to record your mix to. That way, it is possible to listen to final mix before it is bounced, and to keep multiple prints for comparison purposes.

Random tips
- Most important frequency range is between 200 Hz and 1 kHz, which contains fundamentals for most instruments.
- Do not focus on things can only be heard in solo. Solo instruments to correct issues, otherwise apply effects in the context of the overall mix.
- Voice always needs compression, start with an LA2A, possibly to do two compressors in series, e.g., a lot of LA2A then a little 1176.
- Consider side chains, e.g., kick drum as side chain to a bass guitar (or use EQ and cut out 90 Hz on the kick and 90 Hz on the bass).
- Use EQ bracketing to limit frequency bandwidth to what is needed, e.g., remove frequencies below 100 Hz from vocals, and above 6 kHz from electric guitars. Cut both highs and lows from instruments that only provide texture.
- Start drum mixing with overheads – the microphones farther away from the source typically sound more natural.
- Use HPF before and LPF after reverbs, particularly on plates and springs.
- Split bass into frequency ranges and apply chorus on higher frequencies to get a stereo effect.
- Be careful with Exciters, they typically are being used too much, which results in compressed audio.

Basic mix
- Watch overall loudness of the mix and aim for to leave sufficient headroom for mastering.
- Consider a compressor on the mix bus for controlling volume and dynamics.
- Perform volume leveling and panning for individual tracks and groups to place instruments within the stereo field at appropriate volume levels.
- Use filters and EQ to correct issues with any individual tracks.
- Use reverb on drums and vocals (e.g., a plate) to create depth in the mix and put the performance into an acoustic space. This might be more relevant at this stage for a rock or jazz recording vs. an EDM track.
- Once the basic mix is satisfactory, save (print/bounce) the basic mix for later reference.

Final mix
The final mix is the result of an iterative process and may vary greatly depending on personal experience and/or guidance from the artist or producer. The following would be an appropriate sequence of steps:

- Start with a combination of EQ and Compression for individual tracks or groups of tracks, which may include parallel compression, with the goal being to achieve the right balance between instruments to have every instrument audible in the context of the role it plays.

- Subsequently add effects to make things interesting and to keep the listener engaged. Examples include adding distortion, reverb, or delay to individual tracks.
- Subsequently add automation to parts of the mix and/or to elements of the mix again with the objective of maintaining listener engagement. This could include varying the volume level between verses and chorus, more or less reverb and delay during sections of the performance.

Mixing Vocals

The lead vocal is typically the most important part of a performance and plays a key role in engaging the listener. With that, a lot of attention is being put on processing vocals, and everybody might have a favorite vocal chain. A lot depends on genre and trends might shift quickly. The following are suggestions on what to consider:

- Basic Mix
    - Use a plate reverb to position vocals in an acoustic space with the instruments,
    - Cut low-end rumble and muddiness with a HPF and use high-shelf EQ for presence, air.
- Further processing with inserts to taste
    - A compressor (e.g., 1176) to tame the peaks and reduce t's and k's (ratio 4:1, medium attack, fast release).
    - One or multiple EQs to further shape the tone.
    - A leveling compressor (e.g., LA2A, TLA100) for balance, up to -2 dB gain reduction and make-up gain.
    - Additional compression (e.g., SSL, 1176) with 3:1 ratio, fast attack, and medium release for up to -3dB gain reduction
- Further processing with sends to taste
    - Bring the vocal more upfront with parallel compression
    - Add air to help the vocal stand out
    - Add delay and chorus for depth and breadth
- Vocal tuning
    - If vocal tuning is required or desired, it should be done at the end of the vocal processing and in the context of the mix rather than solo/standalone.
    - Melodyne is the industry standard for tuning. Note that Melodyne will create separate copies of vocal tracks, so any edits done after Melodyne will not be captured.
- Automation
    - Automate vocal loudness of individual words or phrases for emphasis and to create and maintain interest. Also check on plosives, and "s", "t", "c", "k" sounds and lower volume on these as needed.
    - Automate reverb, other effects during different sections of the songs to increase variation.
    - Add throws, e.g., short increase of reverb or delay to highlight words or phrases and increase variation.

Mix Bus Processing

Adding effects to the mix bus is a means to ensure the mix "comes together" as a whole. One possible approach is often referred to as top-down mixing where effects are applied to the mix bus first before addressing individual tracks or groups. It could mean less work by addressing some basic issues and/or

opportunities in one place rather than having to potentially touch each individual track; it could also mask issues and/or opportunities that should be addressed at the track or group level.

Mix bus processing can be considered a two-step process, the first aiming towards creating the best sound, the second being a more technical part towards creating a finished, competitive, and loud enough mix. The latter is optional if there is a separate mastering step.

Creative mix bus processing would include:

- Compression to use as "glue" for bringing together
    - Use a master bus compressor, e.g., SSL G-Bus, Focusrite Red, Fairchild 660/670.
    - Should be small strokes, with 2-3dB gain reduction on the loudest part of the song:
        - Use ratio 2:1 (higher will sound bumpier, less chilled)
        - Start with fast attack and medium response and dial threshold to desired gain reduction (with Fairchild increase input before setting threshold)
        - Then slow down attack until drums sound right (will make things sound lighter, 1 to 10ms),
        - Then adjust release so that gain reduction meter aligns with the rhythm of the song (creates natural sound/transparency)
    - Use side chain filter to avoid the compressor responding to low end (use HPF 100 or lower) and find balance that does not separate low-end from the rest.
    - Use make-up gain.
- EQ
    - Use to add color and character, e.g., a smiley-face approach with boosting low and high frequencies with a Pultec EQ boosting and attenuating at 60 Hz and at 10 kHz (boost is a shelf, attenuate a bell).
- Consider using a multiband compressor to fine-tune frequency ranges.
- Consider using other effects with caution, e.g., Dynamic EQs/Compressors/Exciters (e.g., Soothe, Smooth Operator, Gregg Wells Mix Centric), or a stereo widener (e.g., Izotope Imager)

Technical mix bus processing is described in the mastering section below.

Some PLAP Resources
- [Anatomy of a mix](#)
- [Pro Tools basics - mixing](#)
- [5 Essential mixing tips](#)
- [LCR panning and mixing in mono](#)
- [10 Things to do in every mix](#)
- [5 Vocal mixing tricks](#)
- [5 Quick master bus tricks](#)

Mastering
Mastering is traditionally a mostly technical discipline for preparing recordings for the publishing medium or media (e.g., tape, vinyl, or CD). Mastering ensures that the recording will support the technical specifications and constraints of the medium. Mastering also helps with ensuring consistency e.g., across the tracks of an album in terms of loudness and sound.

The following is a rather superficial perspective on mastering a recording, and the actual mastering work for a release is preferably done by a mastering engineer. Automated mastering services are legitimate alternatives with pros (e.g., cost) and cons (e.g., limited options/flexibility). Here is a list of steps to consider for mastering:

- Cleansing: Perform a cleansing on the mix using Izotope RX or equivalent to remove any left-over unwanted noise.
- Saturation: Experiment with caution to potentially add energy to the mix.
- EQ:
  - Use with caution to improve the overall sound of mix, less is generally more. Moves should be small in 0.25dB increments, and use wide Qs are they are more musical
  - Consider
    - High shelf for room, airiness
    - Low shelf to add warmth
    - 8 kHz boost to add presence
    - 125hz boost for body
    - 250hz boost for warmth
- Compression:
  - Ratio: Fairly low and gentle
  - Attack: Set long enough to let transients come through, will make things sound bigger, punchy, and more present. For sharper, snappier transients, an attack time of 10–30 ms range might work. Fast attack times can push an instrument back in the mix.
  - Release: Slower release times will smooth out dynamics and can be used to push an instrument back in the mix / control its depth. Faster release times add excitement, energy, and a perception of loudness.
  - Threshold: Usually high because ratios are low
  - Knee: Soft knees have a smoothing effect. Hard knees are more edgy
- Limiting:
  - Input gain is the amount of gain that feeds into the limiter. The more gain, the more gain reduction, the louder master.
  - Set output gain limit somewhere between -0.2 dB$_{FS}$ and -0.02 dB$_{FS}$.
  - Ideally the limiter is transparent and gain reduction stays below 2.5dB.
  - Too much limiting moves low frequencies backwards, too fast release can lead to distortion.

PLAP Resources
- [The ultimate guide to mastering](#)

Tools

Filters and Equalization (EQ)
- Filters cut/attenuate frequencies within an audio signal:
  - High Pass Filters (HPF): cut frequencies below cut-off, let frequencies above pass.
  - Low Pass Filters (LPF): cut frequencies above cut-off, let frequencies below pass.
  - Band Pass and Band Rejection Filters: combination of HPF and LPF

- Controls
  - Center or cut-off frequency: sets frequency to filter from or to
  - Slope in dBs/octave: steepness of the slope, higher means steeper
- Equalizers cut or boost frequency ranges in bell or shelf form
  - Controls
    - Center Frequency: which frequency to cut or boost
    - Slope (Q): steepness of bell or shelf
    - Amount (dB): to cut or boost
  - Frequency ranges:
    - Absolute: Sub (20-60), Low (60-100), Low-Mid (100-2k), High-Mid (2-6k), High (6-20k)
    - Relative: Weight, Body, Tone, Definition, Presence, Clarity
- Using EQ
  - Use separate EQs for corrective (cutting) and for enhancing (boosting) and start with corrective, e.g., during the basic mix:
    - Corrective (technical): remove annoying frequencies, using transparent EQs
    - Enhancing (creative): highlight instrument sweet spots, using EQs that color
  - Cut low/high frequencies aggressively for texture instruments, but not for main instruments.
  - Use EQ during recording only if sound cannot be improved with mic placement or other techniques.

Compression
- A compressor is an automated volume control that raises soft sounds in relation to loud sounds and reduces dynamic range, as in the delta between the softest and the loudest sound.
- Controls
  - Ratio is the amount of compression applied once the signal reaches the threshold:
    - Compression: (<10:1): shape sound
    - Limiting (>10:1): achieve an overall sound level e.g., during mastering, by aggressively reducing sound that exceeds the threshold level, and
    - Brickwall Limiting (Infinity:1): cut off sound that exceeds the threshold level.
  - Make-up gain: since compression will reduce the volume of the signal, use make-up gain to return signal to original volume.
  - Attack: how fast will compressor react once the signal reaches the threshold
  - Release: how fast will compressor release once the signal drops under the threshold
  - Knee: soft or hard transition from original to compressed signal, e.g., use hard knee for transient-heavy sources
- Using upward (or parallel) compression
  - Creating a copy of the audio signal for compression and combining the un-compressed and the heavily compressed copy of signal, typically prior to adding other effects to the then combined signal.
  - Use for vocals to make soft parts louder without touching the louder parts.
  - Use for drums use for kick and snare (individually) and add to drum bus.
  - Consider 1176 compressor in "nuclear" (all ratios pressed) mode for this.

- Types of compressors

Type	Construction	Characteristics	Examples
VCA (Voltage Controlled Amplifier)	Integrated circuits	Very fast attack/release, most flexibility/versatility, very transparent, clarity for level control, but no coloring. Console compressors are typically VCA, use on Master Bus, and for transient control, e.g., on a snare.	SSL master bus or channel (punchy/tight) Neve channel and 33609 master bus (thick/smoother) API 2500, dbx-160
FET (Field Effect Transistor)	Level automation by transistors	Very fast attack and release, aggressive, crunchy, coloration (add harmonics)	1176 & 1178: fix threshold based on ratio, attack/release 1 (slow) to 7 (fast) add color, for parallel compression, distortion, not for leveling
Opto (Optical)	Input triggers photo resistor based on brightness	Slow attack, release auto or slow, used for levelling compressors	LA-2A, LA-3A, CL1A, CL1B - Musical, smooth, let transients pass; typically, not used on mix bus
Tube	Tubes regulate gain reduction	Smooth, thick, creamy, use for glue, mostly or master busses, lessen depth	Manley Vari-Mu (Master Bus), Fairchild 660, 670 (Master Bus), TLA 100

Effects

Reverb

- Reverb components
  - Direct Sound: 0db in 0 sec (=reference)
  - Pre-Delay: gap between direct sound and early reflection, depends on size and position of the sound source in a room, gives sense of space and distance from the source
  - Early reflections: delayed first copy of the direct sound reflecting from borders
  - Late reflections: additional reflections following the early ones, also referred to as "tail"
- Types of reverbs
  - Echo chamber (most natural)
  - Mechanical: Spring and plate (no pre-delay)
  - Digital hardware
    - Algorithmic: based on math – room, hall, cathedral, etc. (e.g., Lexicon)
    - Convolution: based on impulse response (I/R) (e.g., Bricasti)
  - Digital software
    - Algorithmic, convolution, emulation, …
- Notes
  - When using reverb in mixing, use a send for blending in reverb; when using reverb during recording use an insert and reverb becomes part of the source sound
  - For creating depth in a mix use room, or plate with pre-delay
  - Plate: shimmers more than room, and is good for vocals and percussion because plate retains transients very well vs. getting lost in a room reverb; use to increase body, make something stand out
  - Spring: creates twangy sounds, e.g., on guitar
  - Room: use to put something in an acoustic space
  - Hall: use for big stadium sound, e.g., 80s romantic, Whitney Houston, Celine Dion

Delay

- A delay is one repeat, a feedback delay are multiple delays, a reverb is an infinite number of delays of a signal in an acoustic space
    - A feedback delay creates a delay and then a send from the delay to itself, which results in feedback
    - An echo is natural delay, used to be equivalent to reverb, now used separately
- Initially, delays were created by using multiple tape machines during recording:
    - Slap-back delay: recording two mics into two tape machines and from there into the mixer. The delay results from the space between record and play head of the second tape machine
    - Artificial Double Tracking (ADT): route signal from one tape machine sync head to a second tape machine, combine signals in mixer and use capstan control to vary tape speed of the second machine.
    - Delay time was originally determined by distance between record head and play head on a tape machine, and the speed of the tape, e.g., 15 ips tapes speed creates 125-175 ms delay
    - Tape delay characteristics
        - Wow and flutter caused by small delays in tape speed (amplitude and frequency)
        - Natural roll off (dampening) by losing high frequencies over time (LPF on a delay plugin simulates that)
        - Increase in distortion caused by noise floor of the tape machine
        - Use delays to blend in (in time with BPM) or to stand out (out of time with BPM)

Other effects

- Comb filter:
    - Multiple delayed signals will result in constructive and destructive interference which on a graph will look like a comb.
- Flanger:
    - Originally implemented by using two tape machines, synchronizing them, then slowing the second tape down manually (put finger on the flange), after which it will speed up to get back in sync and creating variable delay time / comb filter and pitch shift.
    - The ratio between original and "flanged" signal should be 50:50 (otherwise it goes back towards the original signal)
    - Rate = how fast or slow the comb filter moves, how fast the delay time changes
    - Depth = how deep the comb filter will be (= delay time)
    - Flanger plug-ins use LFO (Low Frequency Oscillator) to manage Rate/Depth
    - Sounds natural/unpredictable
- Phaser:
    - Combine original signal and phase inverted frequencies, should also be 50:50
    - Can be used to make cymbals sound less harsh
    - Rate = how fast or slow
    - Depth = how far or close to center frequency
    - Sounds steady/predictable
- Chorus
    - Different copies with different delay times and modulated
    - Chorus FX have rate and depth

- o   Use in all mixes as a tool, e.g., on vocals (2 – left and right) to add movement, make vocals more interesting, not in the foreground, not as obvious
  - o   Use to make something richer (instead of compression)
- Distortion
  - o   When a signal exceeds the maximum operating level (=loudness) of analog equipment it will be smoothed off and with that change its form towards a square wave. That distortion sounds pleasing to the ear and is therefore desired. Guitar amplifiers typically create warm distortion, transistor amplifiers create harsher distortion.
  - o   Distortion is also called saturation, referring to the increase in harmonic content
  - o   High frequency distortion will bring sound closer
  - o   Examples: Softube Saturation Knob, Lo-Fi (Pro Tools stock plug-in), FabFilter Saturn, sending a signal through a Guitar Amplifier plug-in
  - o   Distortion is not desired in the digital world since signals that go above the sample rate limit are rounded to the upper limit. This is referred to as clipping and creates harshness.

Some PLAP Resources

- [5 Quick EQ tricks](#)
- [5 Quick compression tricks](#)
- [3 Delay mixing tricks](#)
- [5 Quick delay tricks](#)
- [5 Quick reverb tricks](#)