Digital Sound Capstone
DXARTS 460

Lecture 5:
Processing, mixing and mastering

Spring 2011

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Composing, processing mixing and mastering

- In most popular music genres the processes of composing, recording, mixing and mastering are usually separated, often happening by different people altogether
- In electronic/electroacoustic music, there is very often much crosstalk between them, with one seeping into the other.
- Processing, mixing and mastering are the different stages of sculpting sound
- The digital musician should know about this all, and regard it from an artistic-centered viewpoint

Processing| modulation

Modulation effects

- **Tremolo** `Human rate' amplitude modulation
- **Ring modulation** Multiplying two (potentially complex) signals with each other

Processing| Filtering

Filtering

Filter types:

• **Low Pass**	Remove high frequencies (with n dB per octave)
• **High Pass**	Remove low frequencies (with n dB per octave)
• **Band pass**	Only let a certain frequency range through (n dB per octave) A combination of low-pass & high pass
• **Band Reject**/stop (**Notch**)	Attenuate a certain frequency range (with n dB per octave)
• **Low-Shelf**	Increase or attenuate lower frequencies below
• **High-Shelf**	Increase or attenuate higher frequencies
• **Peak EQ**	Make a peak or dip in the spectrum (like adding a band-pass, band-reject to the original)

Parameters

• *Cut-off frequency*	Frequency when filtering start to take place
• *Resonance* or *Q*	The slope of the filter
• *Gain*	Amplitude boost or attenuation

Processing| Delays

Delay-based effects

- **Delay** Playing back a sound with delay, possibly feeding the delayed signal to itself
- **Phasing** Time-varying phase cancellation and reinforcement effects
- **Wah-wah** Time-varying center frequency/resonance for band-pass filter
- **Vibrato** Frequency modulation, by oscillating the read pointer in a delay line
- **Chorusing** Mixing together many delayed copies (1-30 ms delay), with randomly fluctuating delay times
- **Flanging** Combining the signal with a variably delayed copy of itself (delays 0-2 ms with low-frequency oscillation)
- **Echoes** Slapback/doubling (10-30 ms delay) or explicit echoes (> 30 ms delay)

Processing| Dynamics

Dynamic processing: Nonlinear remapping of amplitude

- **Compressor** Acts as an automatic volume control, attenuating loud signals
- **Expander** Boost amplitude of signals above a threshold
- **Limiter** Keep signal peaks from exceeding a preset level. Only affects the highest peaks
- **Noise Gate** Signals with an amplitude lower than a threshold are multiplied by zero, effectively being removed. Useful for removing noise during pauses in a recording

Somewhat related to the last one:

- **De-noising** Techniques to filter signals selectively to remove undesired noise components

Processing| Distortion

Distortion effects

- **Distortion** A nonlinear effect (like dynamics processing) also often based on waveshaping functions; can be used in the modeling of analog components such as valves

- **Bit-rate degradation** Reducing the bit-depth of a signal

- **Sample-rate degradation** Reducing the sampling rate of a signal

Processing | Advanced

Advanced techniques

- **Time stretching** — Breaking down signals and building them back up again in new combinations in time. Commonly used techniques are granulation (time-domain) and phase vocoding (frequency domain)
- **Pitch shifting** — Altering the pitch of an audio signal. Commonly used techniques are granulation, delay based techniques and ring modulation (time-domain), as well as phase vocoding (frequency domain)
- **Harmonizing** — Adding multiple pitch-shifted versions of a signal
- **Vocoding** — An example of resynthesis using a simple filter bank; energy envelopes are extracted in each band (the analysis), and then used to control a bandwise sound
- **Sound morphing** — Interpolating between different source data sets or states through the parameters of an analysis-resynthesis model

Processing

Don’t overdo it!

Mixing (traditionally)

- Putting the different tracks together
- Fixing relative levels
- Spatial positioning (panning)
- Equalization per track/group/bus
- Dynamics processing per track/group/bus
- Effects processing

Mixing

- Things to remember:
  - Sound waves are additive or subtractive, depending on phase
    - Beware of clipping!
    - Beware of comb-filtering
    - It's a good idea to group material that sounds similar together (on the same channel, group or bus), so that you can apply the same processes, and split different material to different groups to make sure you can treat them differently

Mastering

'Making good sound is like preparing good food. If you overcook, it looses its taste'

Mastering engineer Bob Katz, in *Mastering Audio*

Mastering

Mastering workflow

- Listening
- Editing
- Clean-up
- Leveling / panning adjustments
- Processing (first time DSPs should be introduced)
- Rendering

Mastering axioms

- Mastering is an art of compromise
- Every action affects everything
- Once again: Don't overdo it
- Surgical changes are usually better than a brute-force attack
- Don't be satisfied with presets – tweak, listen, judge
- Different music styles and different pieces need different mastering strategies
- Listen to the levels your piece is supposed to be listened at
- Take care of your mastering / monitoring space

Mastering axioms

- Listen through many different sound systems, but work on the best (cleanest) one you can
- Listen carefully before mastering; while mastering; after mastering
- Take breaks and come back to the piece. You cannot master in one day
- Use analysis tools (time/frequency domain) for help, but let your ears be the final judge.
- Keep in mind that while analysis tools tend to show you the peak amplitude, our ears listen to the average amplitude (loudness)
- Listen without looking at the screen if you really want to listen right
- Show your work to others before you call it done

Mastering axioms | A/B testing

- Loudness has a great effect on judgment (louder seems better and clearer)
- A/B testing: comparing *before* and *after* a Digital Signal Process (DSP) has been applied
- Make a lot of blind A/B tests, at *equal loudness levels:* many times the added DSP will actually worsen the sound, making it only louder, therefore fooling you to think it sounds better
- Sometimes, you can only really tell what you're doing if you listen to the whole thing, or at least larger chunks than what you're specifically concerned with.
- Many times you will only notice how something feels *after longer exposure,* especially when the differences are subtle
- Louder sounds better in the short term, but more dynamic range sounds better in the long term (less fatiguing). Find the best solution for your piece

Equalizer (EQ)

- An equalizer is a collection of filters in series
- The standard filters for EQ:
  - Parametric => Surgical
  - Shelving => Overall tonality

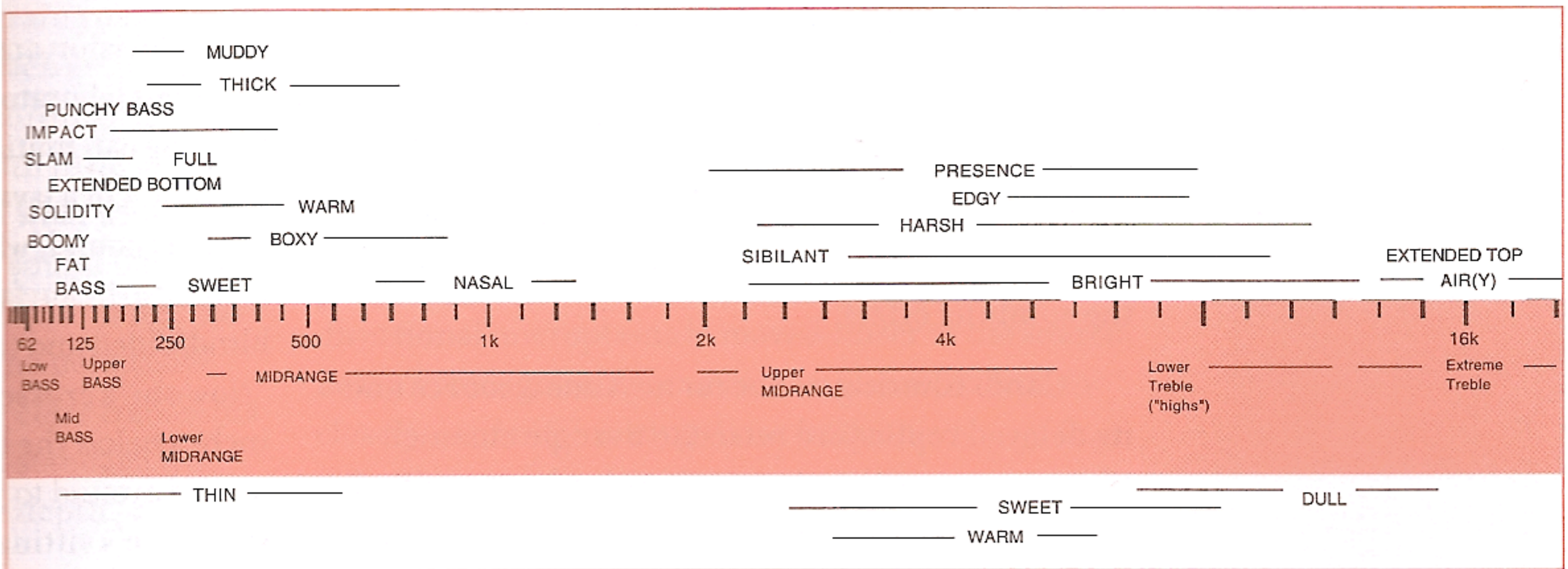
  Parameters: bandwidth, level, Q

EQ axioms

- Try to EQ things before getting to dynamics (compression/limiting/expansion) changes
- Use multiple bands
- Use EQs while mixing, per channel or group, but also while mastering (and don't overdo it)
- Listen carefully for phasing effects
- Work with small increments: 1/2 or 1/4 dB can do the trick!
- Start with the midrange
- Start EQing loud passages first - that's where you'll notice EQ peaks more

EQ axioms

- Applying EQs in one part of the spectrum will affect another. E.g.
  - a slight dip in the midrange (~250Hz) can boost the presence range (~5kHz)
  - Boosting the midrange (~250Hz) can reduce harshness, as can a dip around 6-8kHz
  - adding lows may make your sound duller, reducing may make it brighter
  - adding extreme highs may make it thinner in bass/low mid-range and vice-versa



EQ axioms

- Gentle EQs (with less resonance) sound more natural, as radical slopes cause phasing. Though, you may have to use those, e.g. to get rid of specific feedback resonances

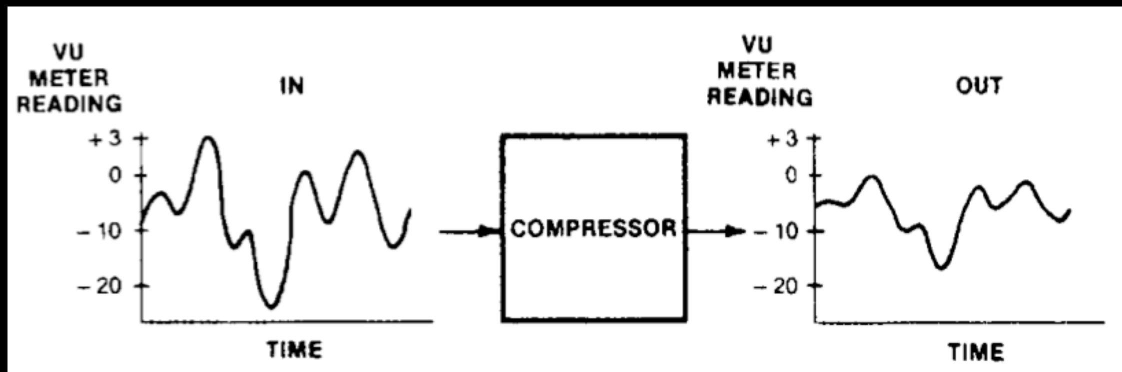
- To find and attenuate a resonance start with a large boost (not cut!) and sweep through the spectrum until you find it. Then, narrow the Q and dip the amount necessary

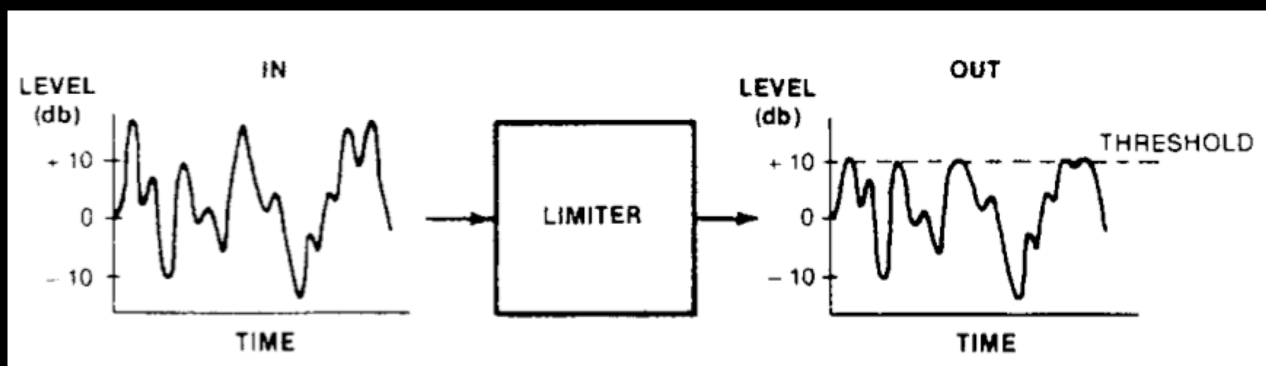
- Excessive subsonic frequencies: Do you really need that 14Hz tone? If not, better remove it, as it may drain the energy from your speakers

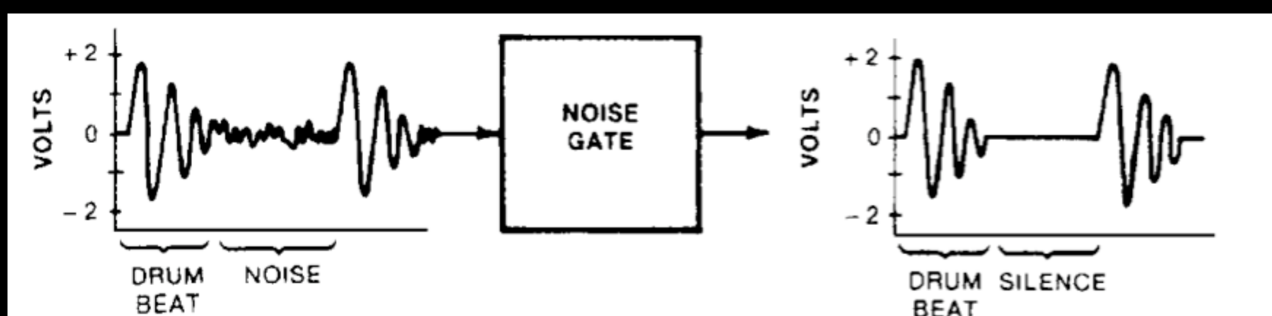
Dynamic processing | Dynamics

- Microdynamics vs meso-/macrodynamics:
  - Meso-/macrodynamics: dynamic differences between passages
    Adjust using amplitude envelopes
  - Microdynamics rhythmic expression
    Adjust using dynamic processors (compressors, expanders, limiters)

Dynamic processing | Processes







Dynamic processing | Dynamics

- Before you do anything with an extra DSP process, make sure you 've done *everything you can* to fix and improve things with plain vanilla *amplitude enveloping* - even if you have to be surgical. This will always sound better

- Equalize before compressing. Do you still need to compress?

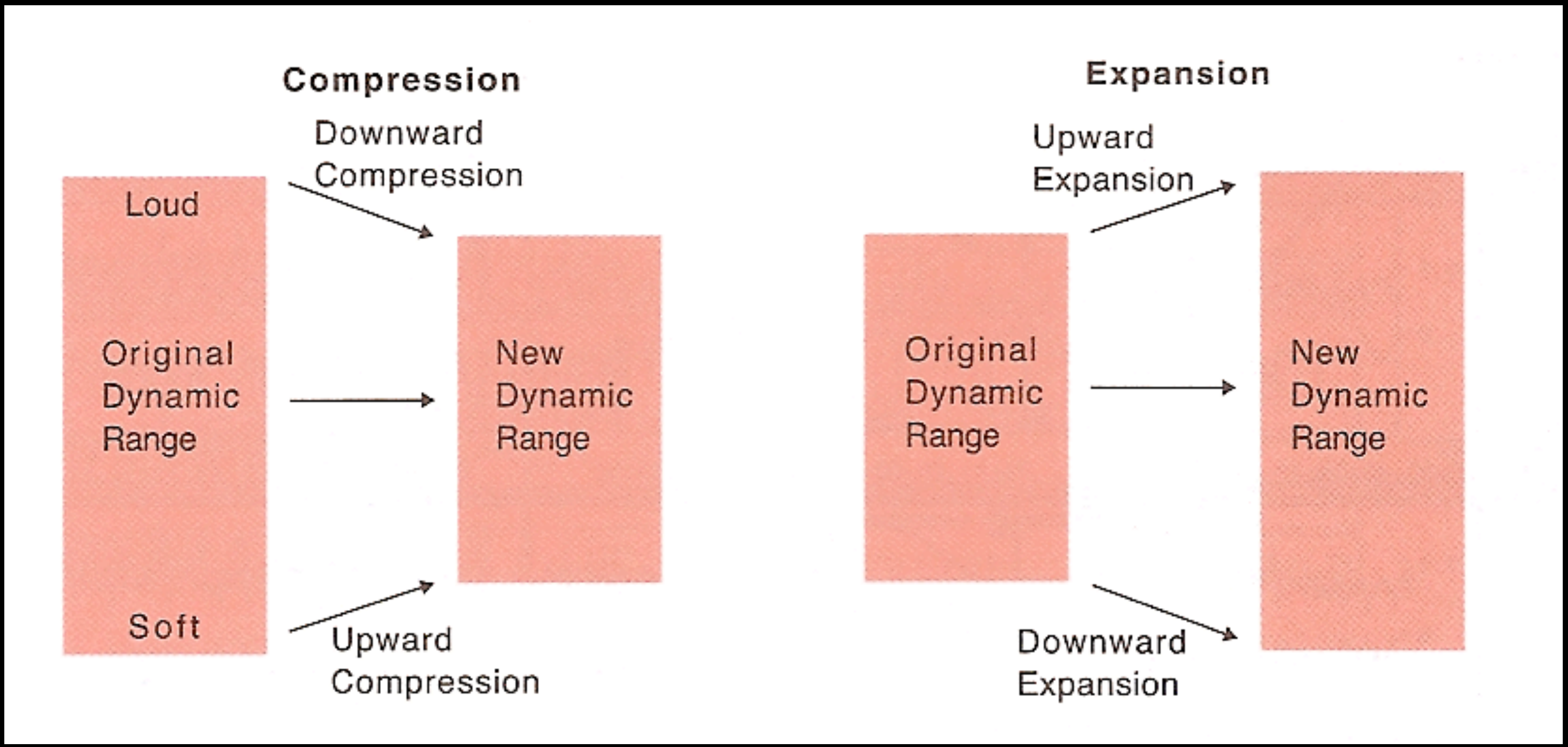
- Compression brings things to the front, but reduces space and depth: is that what you want to do?

- Always try to start with good sounding material. 'Fixing it in the mix/master' is a myth. Usually it is better to just re-record, re-do, mix again, or find another sound.

Dynamic processing | Compression/expansion

- Compression: Upwards & downwards
- Expansion: Upwards & downwards

- Upward: make soft passages louder
- Downward: soften loud passages



Dynamic processing | How it works

- Dynamic processors employ a transfer function, to modify an input according to a lookup table of values – sort of like a dictionary. This is the same principle use in waveshaping (aka non-linear distortion), except there it is used for amplitude values

- Incoming RMS values are swapped with the appropriate table value. Below is an example for the similarly functioning waveshaping (aka non-linear distortion)

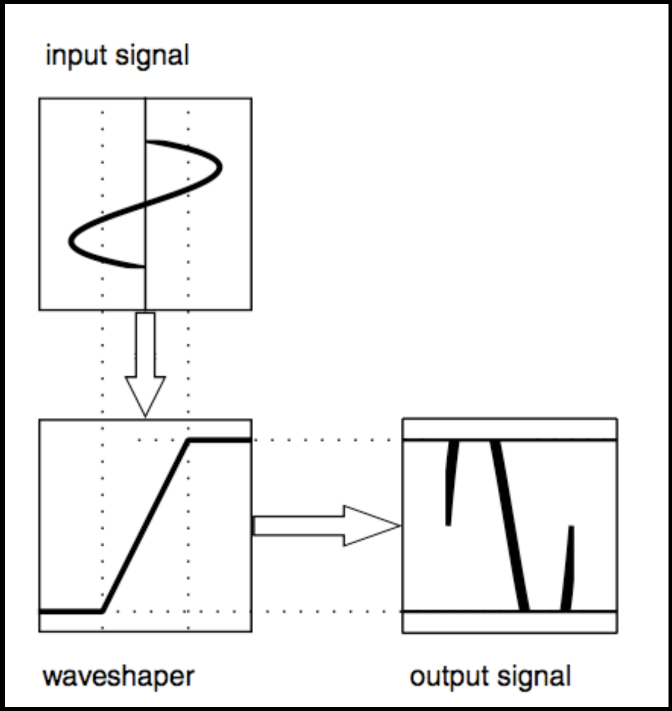


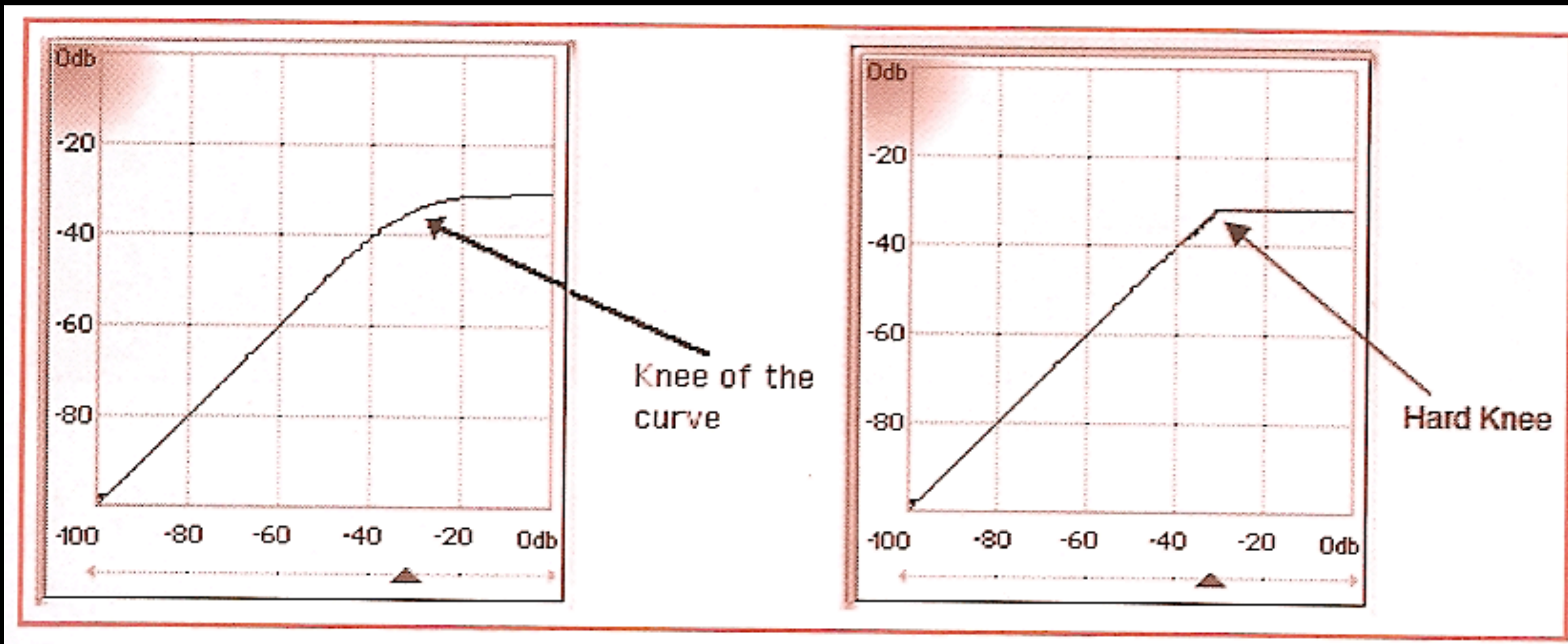
Figure from Miranda, E. R. (2002). *Computer sound design: synthesis techniques and programming*. Oxford, Focal Press.

Dynamic processing | How it works

- **Threshold**: The level when reduction begins
- **Ratio**: The relation between input/output above the threshold
- **Knee**: The portion of the curve near the threshold. Can be soft or hard
- **Gain makeup:** Boosting/attenuating amount to happen to the entire signal after the compression/expansion
- **Attack time:** The time it takes for full gain reduction to occur. Typically between 50-300ms. Avg: ~100ms. Making it too short will start softening attack transients (but maybe that's what you need)
- **Release time:** How long it takes the processor to recover, returning to unity gain. Typically between 50-500ms, or even 1-2 seconds. Avg: ~150-250ms. Too fast a release may produce audible distortion artifacts; too slow will make for a slow recovery
- **Preview/look ahead**: Allows very fast attack times, useful for peak limiting
- **Sidechain**: A signal used to define how strong a compression to apply. Usually this is the same to the processed, but can be a different one. Useful, e.g. dual vs stereo, but also for creative applications

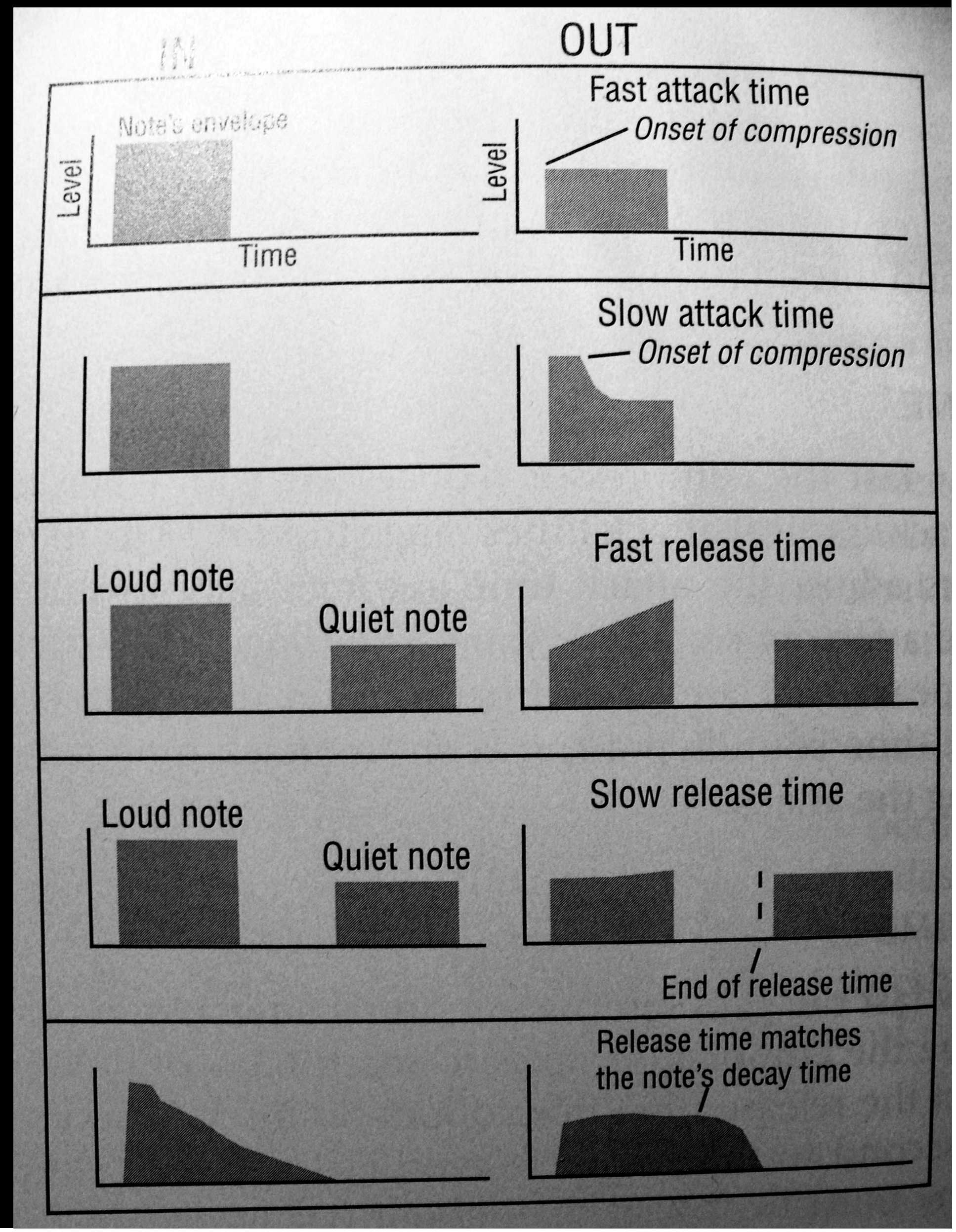
Dynamic processing | How it works

- Soft vs hard knee:



Dynamic processing | How it works

- Effects of attack and release times on a sound



Dynamic processing | Compression/limiting

- Compress when your material seems to lack strength or punch
- Use limiting when you want to raise the overall loudness, without affecting inner dynamics. Also, when you want to minimize the possibilities of clipping
- Don't over-compress: you will loose the dynamic range of the piece
- Be aware that each compressor has its own sound

Mastering | rendering

- **Dither**

  - Only use dither if you're changing wordlenghts (bit-rates), and ONLY in the end of your master. Avoid dithering more than once.

  - Test different dither 'flavors', if available, for each piece

  - Always render uncompressed files. Render a 'master' on the wordlength you're working in the resolution you're working at. If you need a lower resolution file, you can do this using that master file

- **Normalizing:**

  - Finds the highest peak and raises the gain so that it it reaches 0 dBFS (dB relative to full scale)

  - Don't start with it. If your recordings are soft, then you will also make the noise floor very loud

  - Ideally, you won't need to do that in the end of your master, but you can try

Bibliography

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NOTE: All images were taken from these books, unless otherwise noted