Foundations of Audio Engineering:
Audio signal processing - 1

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Partially based on:

- Christopher Ariza. 21M.380 Music and Technology: Recording Techniques and Audio Production. Spring 2012. Massachusetts Institute of Technology: MIT OpenCourseWare, https://ocw.mit.edu. License: Creative Commons BY-NC-SA.
- Digital Audio Production IT3038PA, NITEC Digital Audio & Video Production. 2013. Institute of Technical Education College West.

Audio signal processing

- A signal processor takes an input signal, modifies it, and returns an output signal.
- An audio signal processor can change the a sound signal based on three main properties:
  - Dynamics
  - Frequency
  - Time

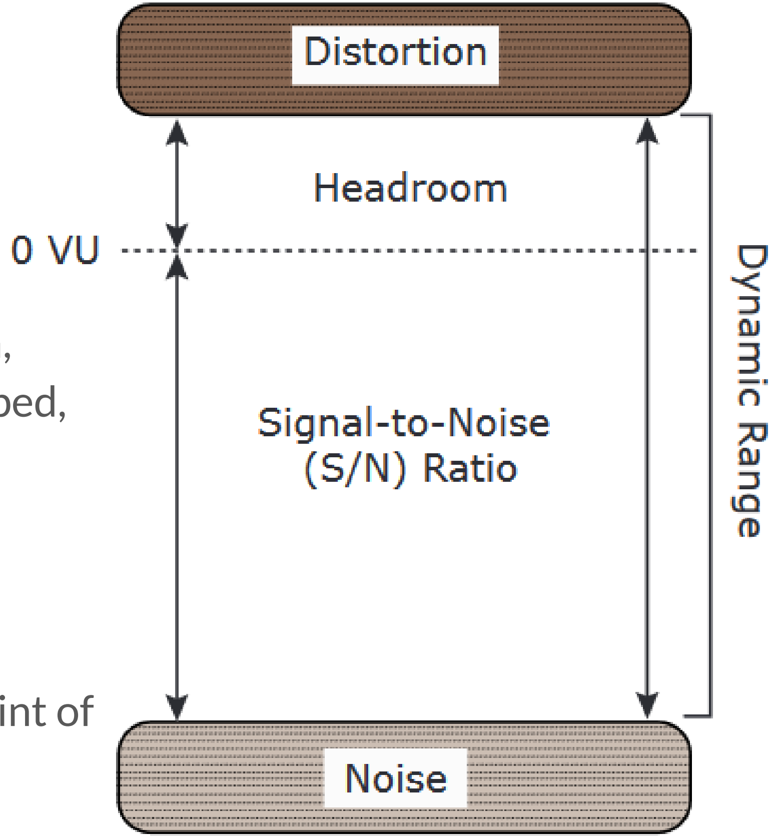
Audio signal processing

- Dynamics
  - Changes in amplitude of a sound, from soft to loud
  - Compressors, Limiters, Expanders, Gates
- Frequency
  - Changes in pitch and harmonic spectrum of sound
  - Equalizers, High-pass filters, Low-pass filters
- Time
  - Some form of manipulation of time to the signal
  - Reverb, Delay, Echo

Controlling Gain

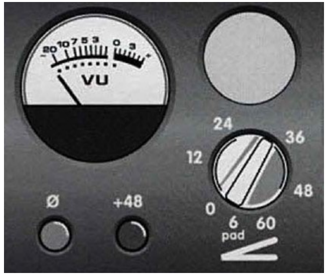
Dynamic Range

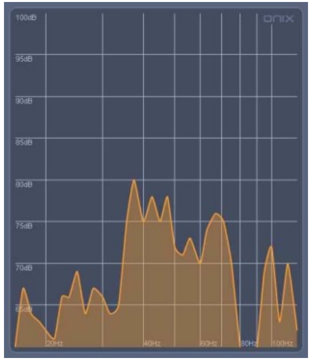
- The range of available amplitudes
- The maximum:
  - peaking, clipping, saturation, distortion,
  - a sine wave becomes square when clipped, and clipping adds harmonics
- The minimum:
  - noise floor
- Operating levels:
  - above the noise floor and below the point of distortion.



Amplitude Meters

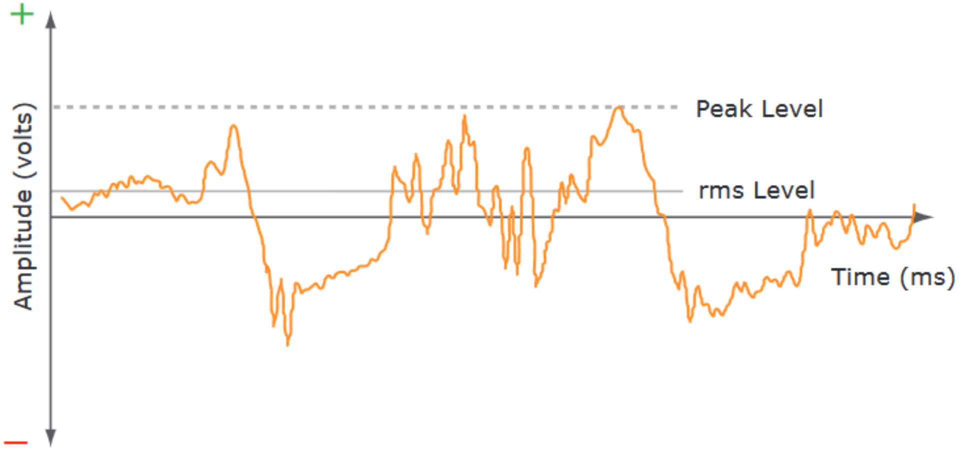
- A simple measure of signals power
- Potentially misleading
- Many varieties
- Considerations when evaluating amplitude meters
    - Peak or average?
    - Units in dB or something else?
    - Negative and/or positive values?
    - Where is 0 dB and what does it mean?
    - What is negative infinity?





RMS Meters

- Root Mean Square (RMS): an average
- Mathematical average
- Average the square of a number (or a window) of samples, then take the square root
- RMS of a square wave is greater than that of a sine wave



dB Meters

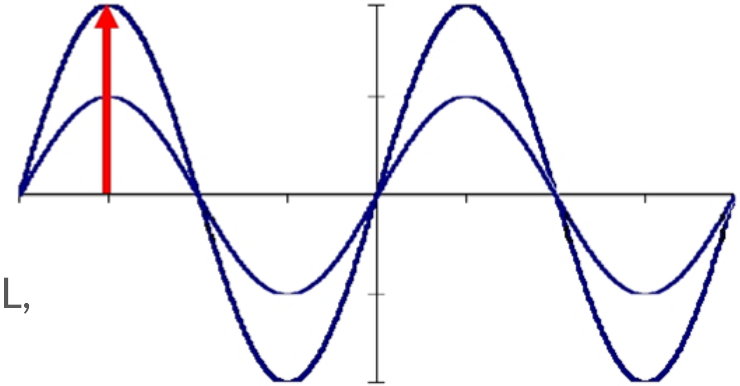
- dBu Meters: negative infinity to +24 dBu
- dBFS Meters: negative infinity to 0 dBFS
- dB SPL Meters: 0 to 120 dB SPL
- Comparisons
    - +4 dBu = -20 dBFS (sometimes -16 to -18 dBFS)
    - -10dBV is equivalent to -7.8 dBu

dBFS	dBVU	dBu
0	6	10
-1	5	9
-2	4	8
-3	3	7
-4	2	6
-5	1	5
-6	0	4
-7	-1	3
-8	-2	2
-9	-3	1
-10	-4	0
-11	-5	-1
-12	-6	-2
-13	-7	-3
-14	-8	-4
-15	-9	-5
-16	-10	-6
-17	-11	-7
-18	-12	-8
-19	-13	-9
-20	-14	-10
-21	-15	-11
-22	-16	-12
-23	-17	-13
-24	-18	-14
-25	-19	-15
-26	-20	-16
-27	-21	-17
-28	-22	-18
-29	-23	-19
-30	-24	-20

created by Murat Böyük Pilavci
stardustmedia

http://www.sengpielaudio.com/calculator-db-volt.htm

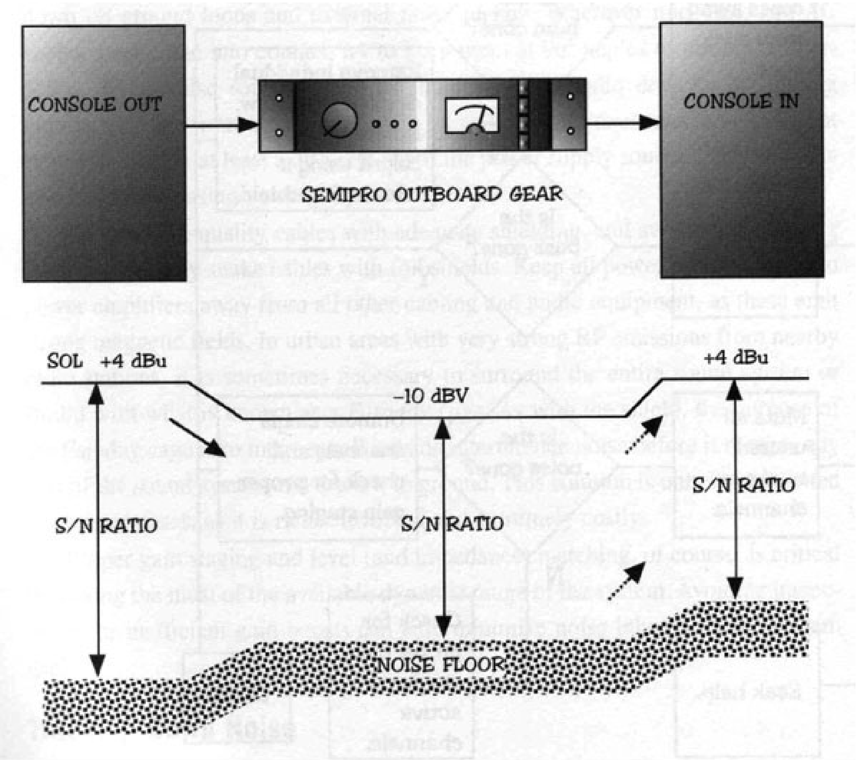
Changing Amplitudes



- Pre-amp (trim):
  - for bringing very quite signals up to SOL, wide range of gain (0 to 60 dB)
- Power amp:
  - for a taking a signal from SOL to a high-powered signal to drive speakers
- Pad (attenuator):
  - reduces gain by a fixed amount with a switch (-6 dB, -20 dB)
- Fader:
  - scales a signal at SOL: unity (no change), boost +10 dB, attenuate to -infinity dB
- Direct Box:
  - convert from -10 dBV to +4 dBu

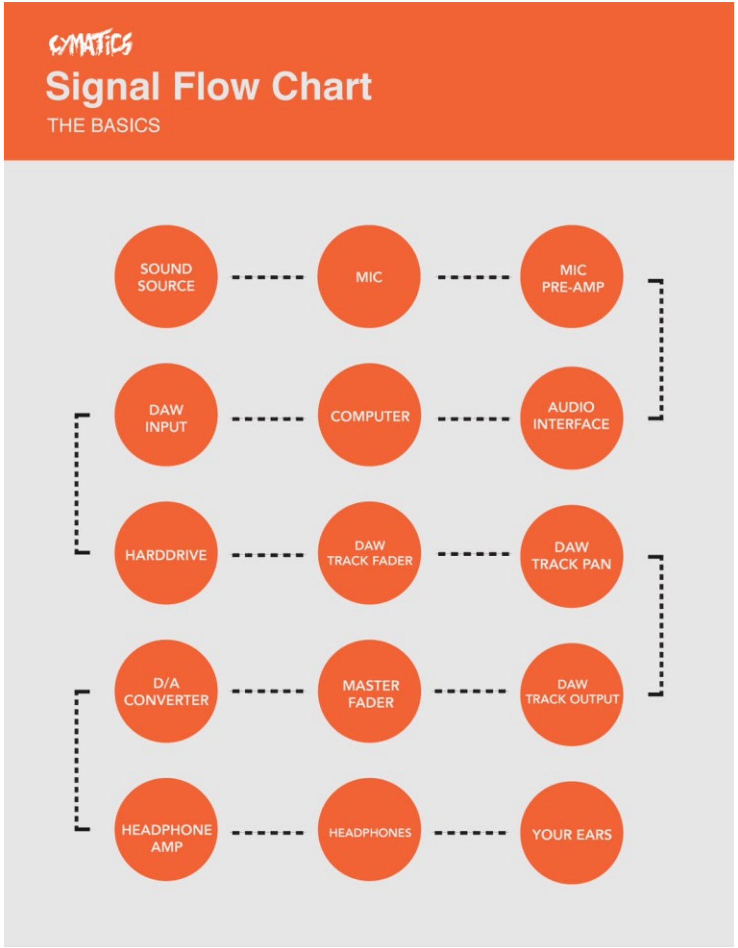
Gain Staging

- Every signal goes through numerous amplifiers from source to destination
- Each amplifier is a gain stage
- Each amplifier (and any process in between) adds noise (has its own noise floor)
- Each gain stage, if above unity, can amplify the last gain stage's noise floor

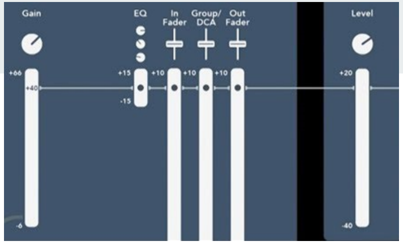


Gain Staging

- Optimal gain staging:
  - first gain stage does all amplification;
  - all subsequent gain stages are at unity
- A device with a poor signal to noise ratio can degrade the entire signal path
- Optimizes signal to noise ratio with ideal gain-staging
- **As much as possible, as early as possible**



Level Setting Procedure

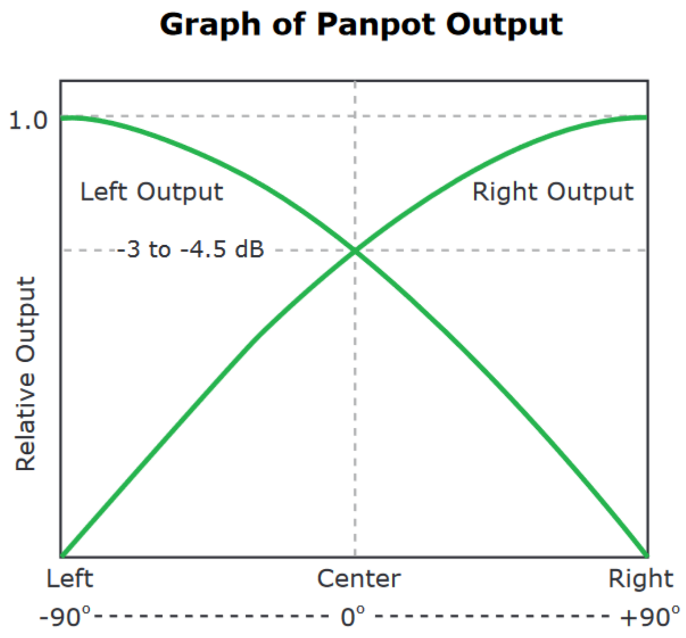


youtu.be/PA0kzFmplMg

1. Reset, clear, and zero all controls *(set trim at minimum)*
2. Connect or select input
3. Set meters to display only the trim gain stage and skip other gain stages
    a. On some mixers, this may mean engaging SOLO
    b. On some mixers, this may mean engaging Pre-fader listen (PFL) SOLO
4. Must get typical material from the source (musician, device, et cetera)
5. Raise the trim slowly
6. Find amplitude peaks and estimate average peaks with meters
7. Continue to raise the trim until average peaks are at +4 dBu (-20 dBFS, 0 VU)

Panning Amplifiers

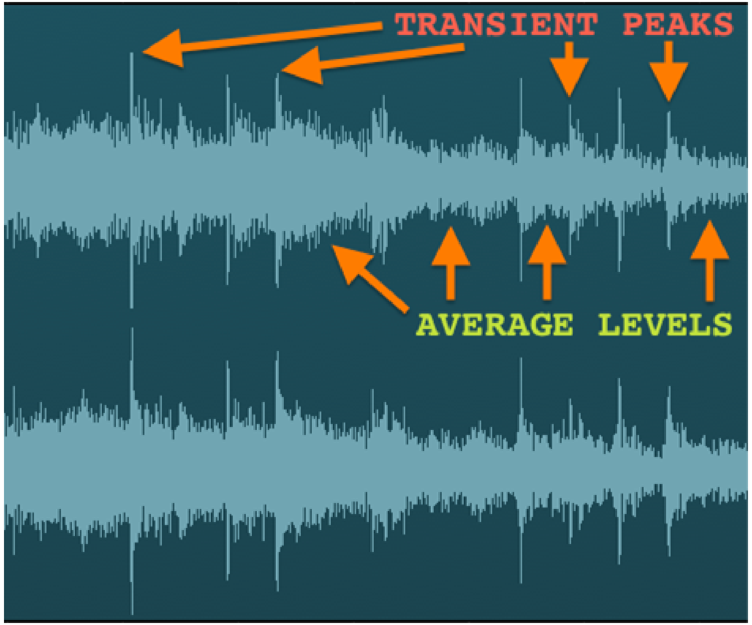
- Linear:
    - Take a signal, split into two signals, and inversely vary amplitudes
    - A fader that as one turns up, the other turns down
    - A bad approach (1 = left, 0.5 = middle, 0 is right)
- Non-Linear:
    - Must reduce amplitude in center to reduce increase in loudness
    - Reduction between 3 dB and 4.5 dB



Dynamic Signal Processors

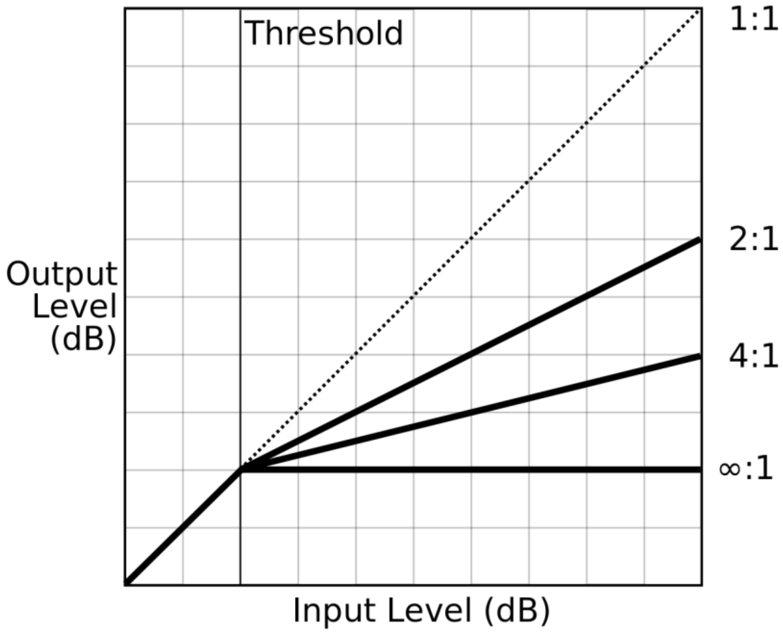
Dynamics

- Amplitude is not the same as perceived loudness
- Perceived loudness has more to do with average signal level (RMS)
- Our ears are more sensitive to amplitudes in certain frequency ranges
- Transients (the attack of instruments) cary essential sonic information



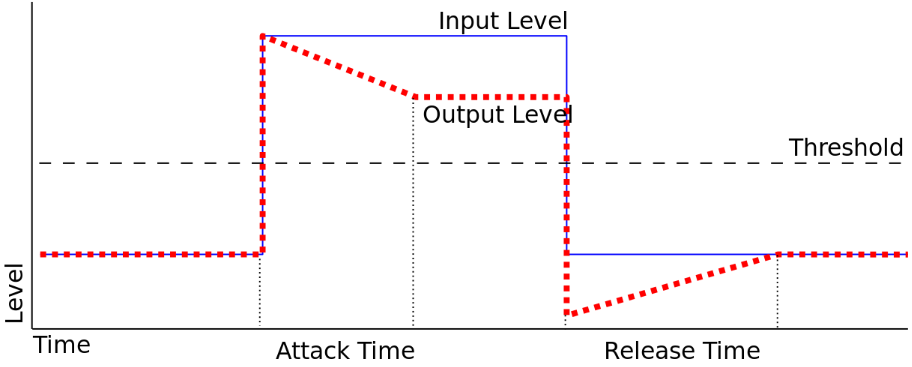
Dynamics Processor Terms

- Threshold:
  - a point of amplitude reference within the dynamic range
- Ratio:
  - used to transform amplitudes by converting input values into output values
  - 2:1 means for every 2 dB in over the threshold, 1 dB comes out
  - 6:1 means for every 6 dB in over the threshold, 1 dB comes out



Dynamics Processor Terms

- Attack:
  - how quickly processing start on onset of amplitude above threshold
- Release:
  - how quickly processing stops on onset of amplitude below threshold

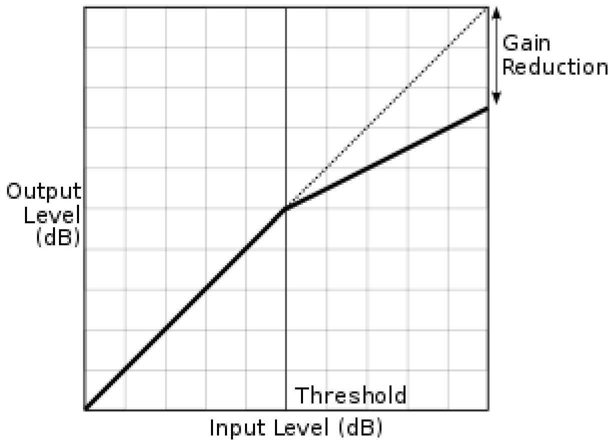


Dynamics Processors

- Two Basic Families
  - Processors that reduce amplitudes when amplitudes are above a threshold (downward compression and limiting)
  - Processors that reduce amplitudes when amplitudes are below a threshold (downward expansion and gating)
- While amplitudes are reduced, this does not mean that dynamic effects only make sounds more quiet

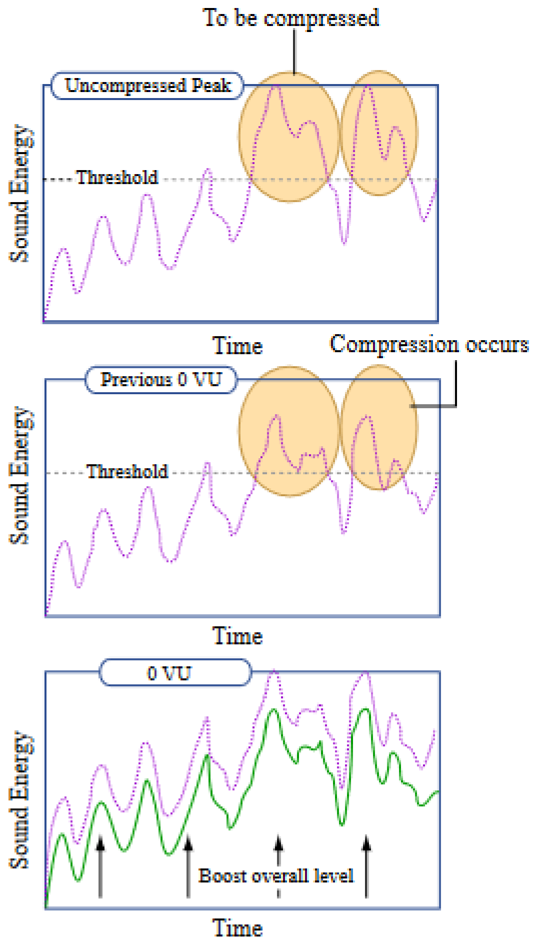
Compressor

- Reduces (compresses) dynamic range and increases average signal level
- Handles situations where a track needs to be turned up but cannot be turned up without clipping
- Often used to reduce the amplitude volatility of a signal: vocals
- Can raise level of quiet signals: can increase sustain, background, and ambience
- Can increase leakage and noise floor



Compressor

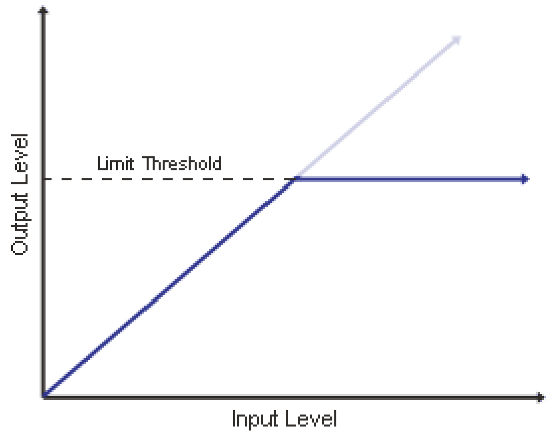
- Compressors work by proportionately reducing a signal's peak volume when it exceeds a maximum level set by the user.
- It leaves anything below that maximum level untouched.
- Two steps of compressors:
    1. Reduce gain above a threshold with a ratio
    2. Increase gain of the modified signal



Compressor

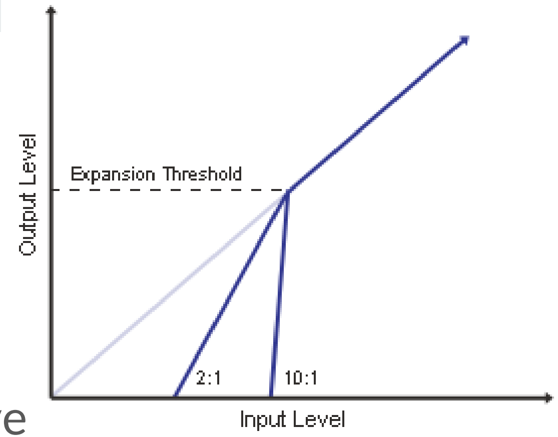
- Attack times generally around 20-50 ms
- Release times generally around 100-300 ms
- Slower attack times are critical for letting transients pass unaffected
- Fast attack times can result in lifeless and unnatural percussion sounds
- Slower release times continue to reduce gain of sustain of instruments
- Pumping: attack and release are too fast and compression is audible; sustain of a signal fades in and out after attack of louder signals
- Breathing: hearing the noise floor slowly rise after the signal falls below threshold; remove by decreasing release time

Limiter



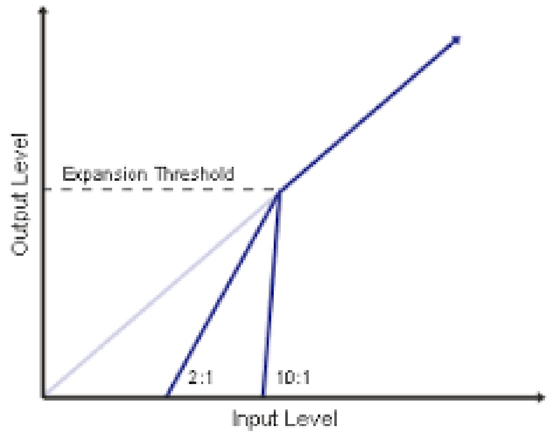
- A compressor taken to an extreme ratio
- Ratios are in the range of 10:1 to infinity:1
- Flattens the top of amplitudes (generally) without distortion (depending on attack)
- Often used to protect equipment and limit dynamic ranges
- Limiters are not able to respond instantly (at 0ms), and so to prevent very rapid peaks getting past the Limiter, most Limiters have a Look Ahead control that “sees” the peak coming.

Expander



- Where a compressor reduces dynamic range, an expander increases dynamic range
- Where a (downward) compressor operates above a threshold, (downward) expanders operate below a threshold
- Expanders makes quiet sounds appear even quieter in relation to loud elements of a signal, because the overall dynamic range has been increased (the gap between the loudest and quietest points).
- Reduce or eliminate leakage, reverb, or noise floor

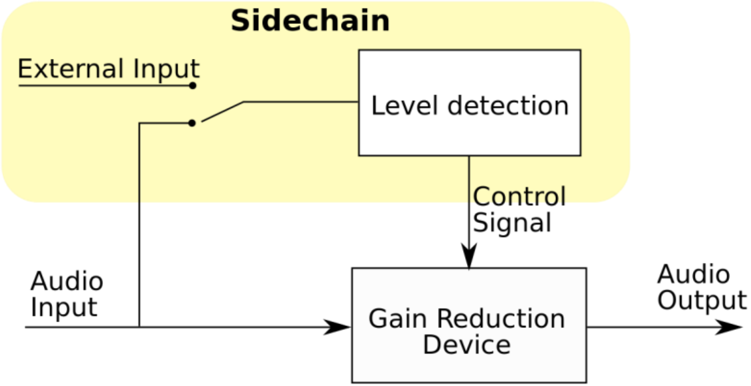
Gate



- An expander at extreme gain-reduction ratio
- Ratios are in the range of 10:1 to infinity:1
- Ratios can be thought of in negative values (below the threshold): -1:-2
- These are Expanders that allow signal levels above a user-definable threshold to pass without any processing, while completely reducing signals that fall beneath that threshold.
- Often used for noise reduction and noise silencing

Sidechaining

- The sidechain is the signal used to trigger the compressor
- Amplitude characteristics of one signal can be used to process the amplitude of a different signal
- Compressor in this case is used without makeup gain



Sidechaining

- Ducking
  - Problem: lower the level of a music track when a spoken voice enters
  - Sidechain is spoken voice or track that needs to be on top of mix
  - Compressor is used without makeup gain
- Deessing
  - Problem: vocals **s** and **th** sounds produce peaks and create extreme presence
  - Sidechain signal is filtered source with boosted problematic frequencies
  - Forces the compressor to react more strongly in that frequency region
- Gating
  - Can use a control track to open and close (gate) on another track

Further learning

- Compressors Explained - https://www.youtube.com/watch?v=IbIC7B4BU6g
- Better Dialogue Audio: Compression and Normalization - https://www.youtube.com/watch?v=9kaI7soRvT0
- Louder Sound with Audition CC: Clip Gain and Compression - https://www.youtube.com/watch?v=PAjpN1Y-anU
- Sound Rescue: Cleaning Up Dialogue Audio - https://www.youtube.com/watch?v=uMAQsq23fII

Practicals