

Digital Audio Processing

EECS 4462 - Digital Audio

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Second level

Third level

F

Fifth level

November 24, 2018

Digital Audio Processing

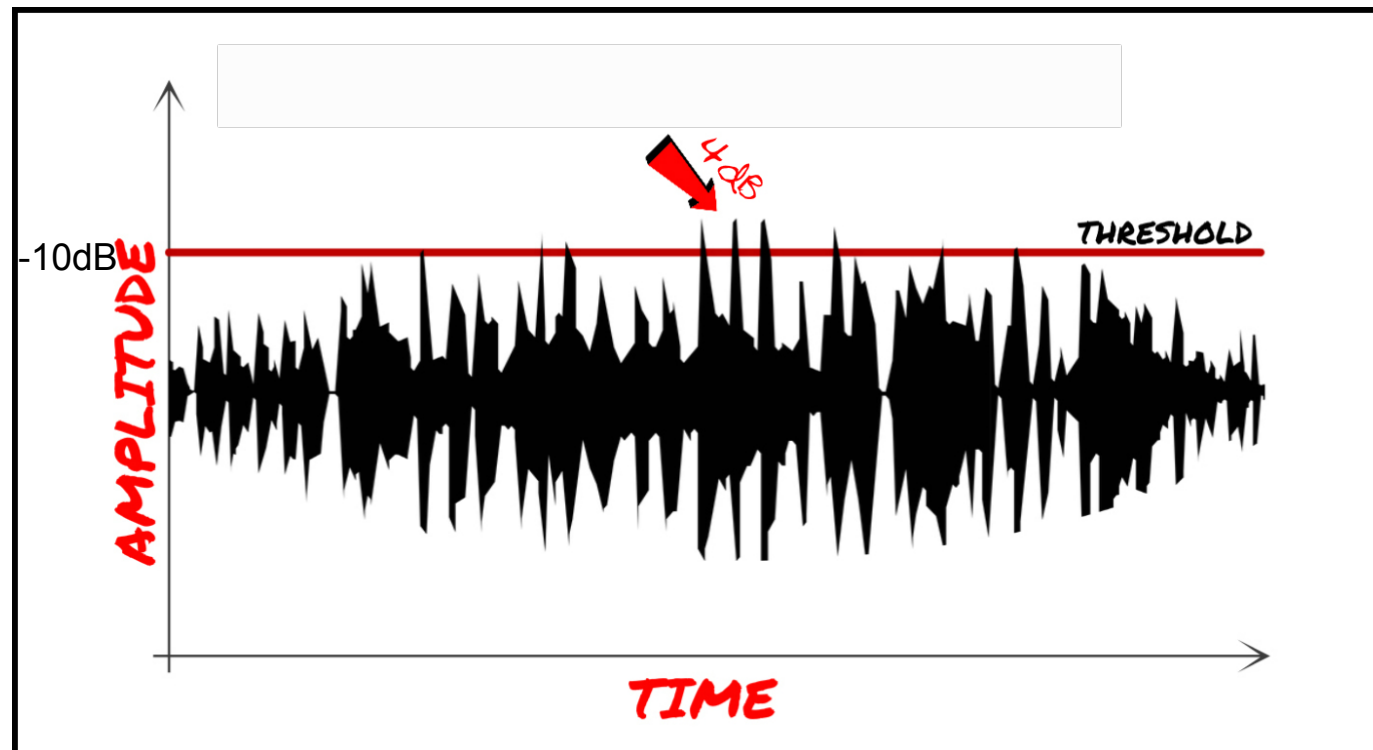
- We have already developed a digital audio processing plugin: a digital delay
- A large variety of other plugins are possible
- **Volume-based:** Gain, Compression, Limiting
- **Time-based:** Delay, Time stretching, Reverse
- **Frequency-based:** Filters, Distortion, Pitch shifting
- **Other:** Reverb, Wahwah, Metering

Volume-based processing

- Adjusting **gain** is rather simple
 - Multiply each factor by a gain factor
 - Care must be taken not to clip
- **Dynamic range compression** is a very common processing operation for both speech and music
- The dynamic range of a piece of audio is the difference in volume between the loudest part and the softest part of the signal
- A compressor makes that difference smaller by making the louder parts softer

Compressor Settings: Threshold

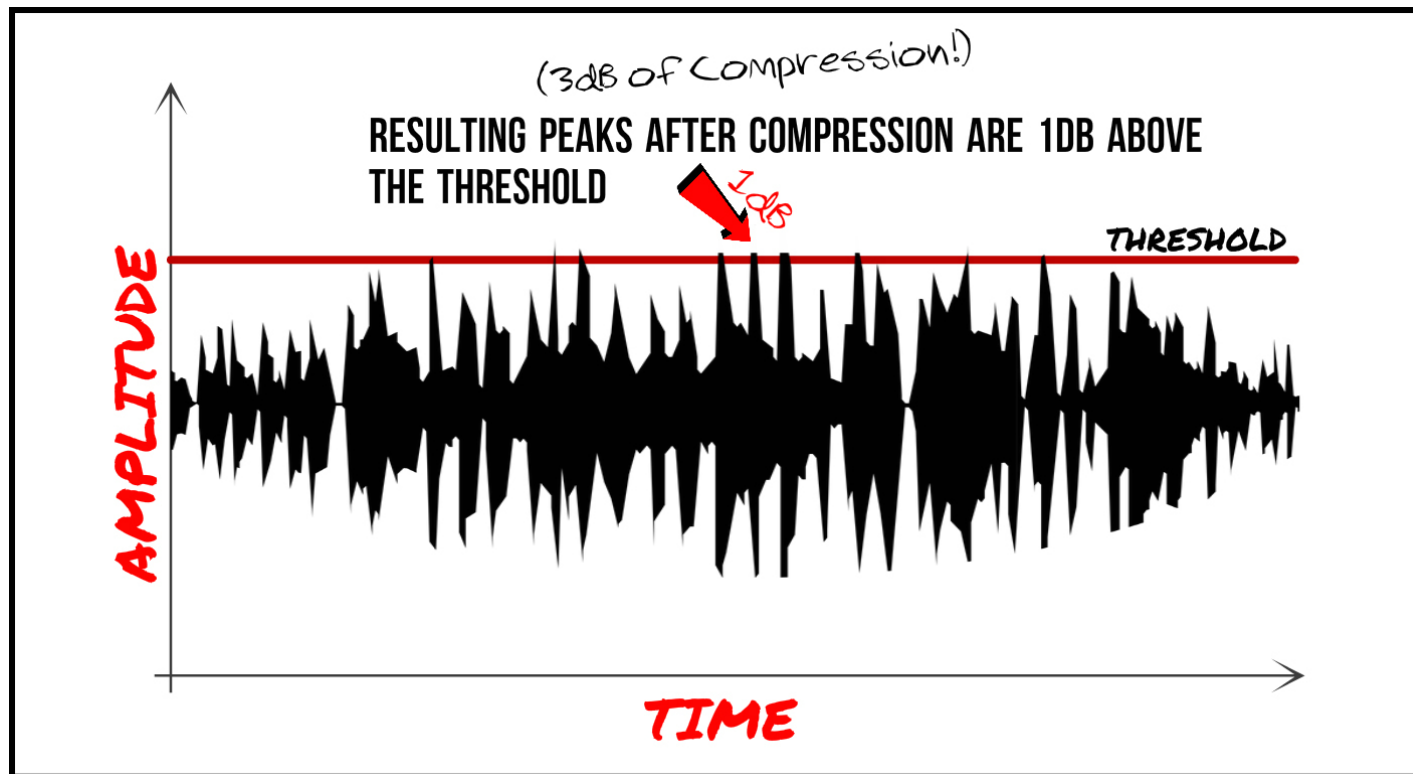
- The minimum level at which compression starts
 - E.g. everything above -10dB must get quieter



- Bottom peaks will also be reduced (not shown)

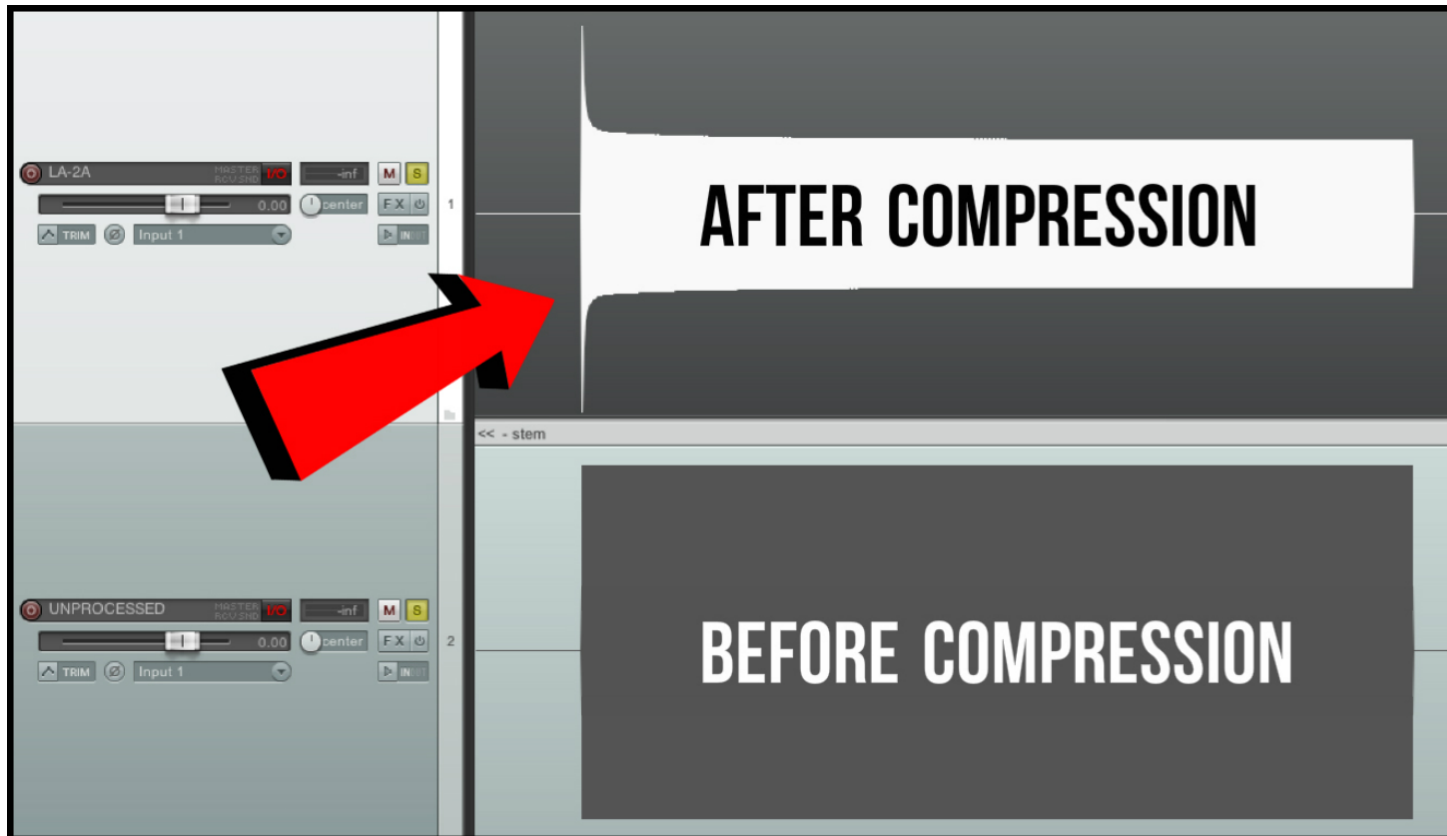
Compressor Settings: Ratio

- The amount of compression applied
 - A ratio of 4:1 means that for every 4dB of level above the threshold, only 1dB will remain



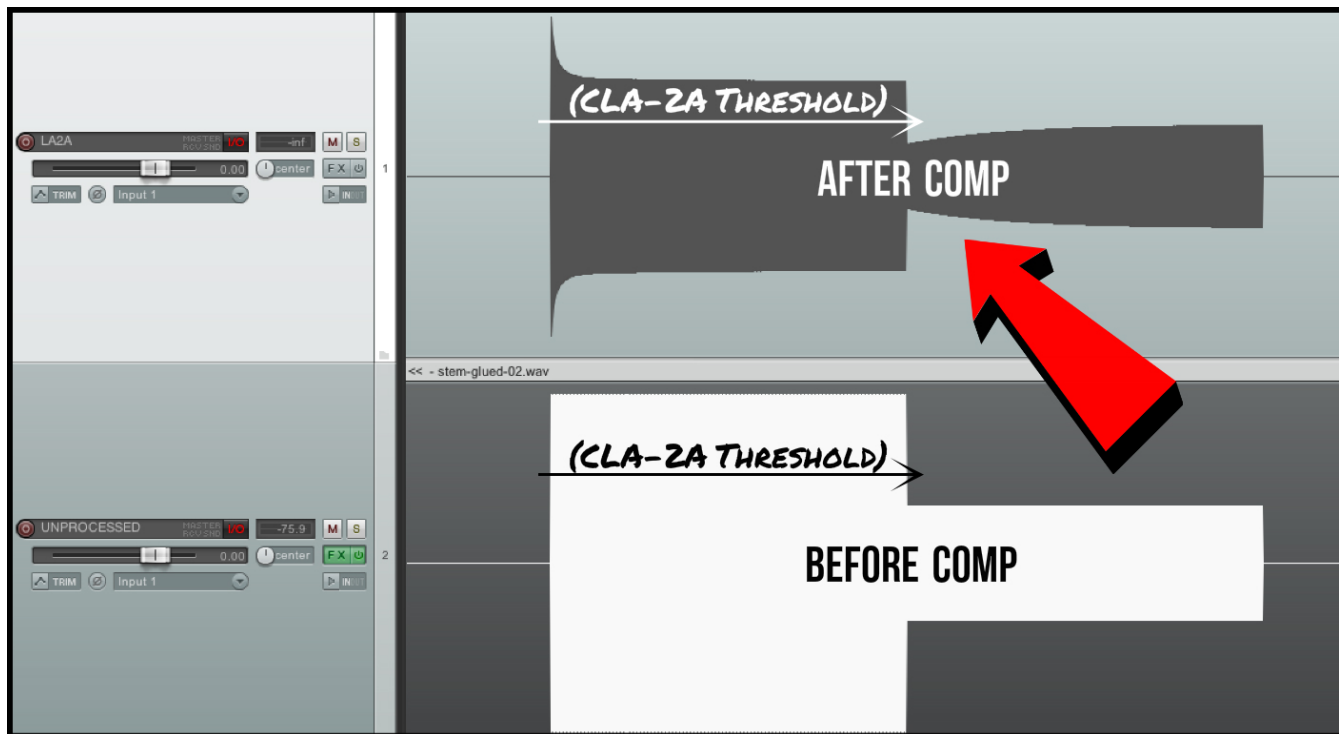
Compressor Settings: Attack

- How fast compression starts



Compressor Settings: Release

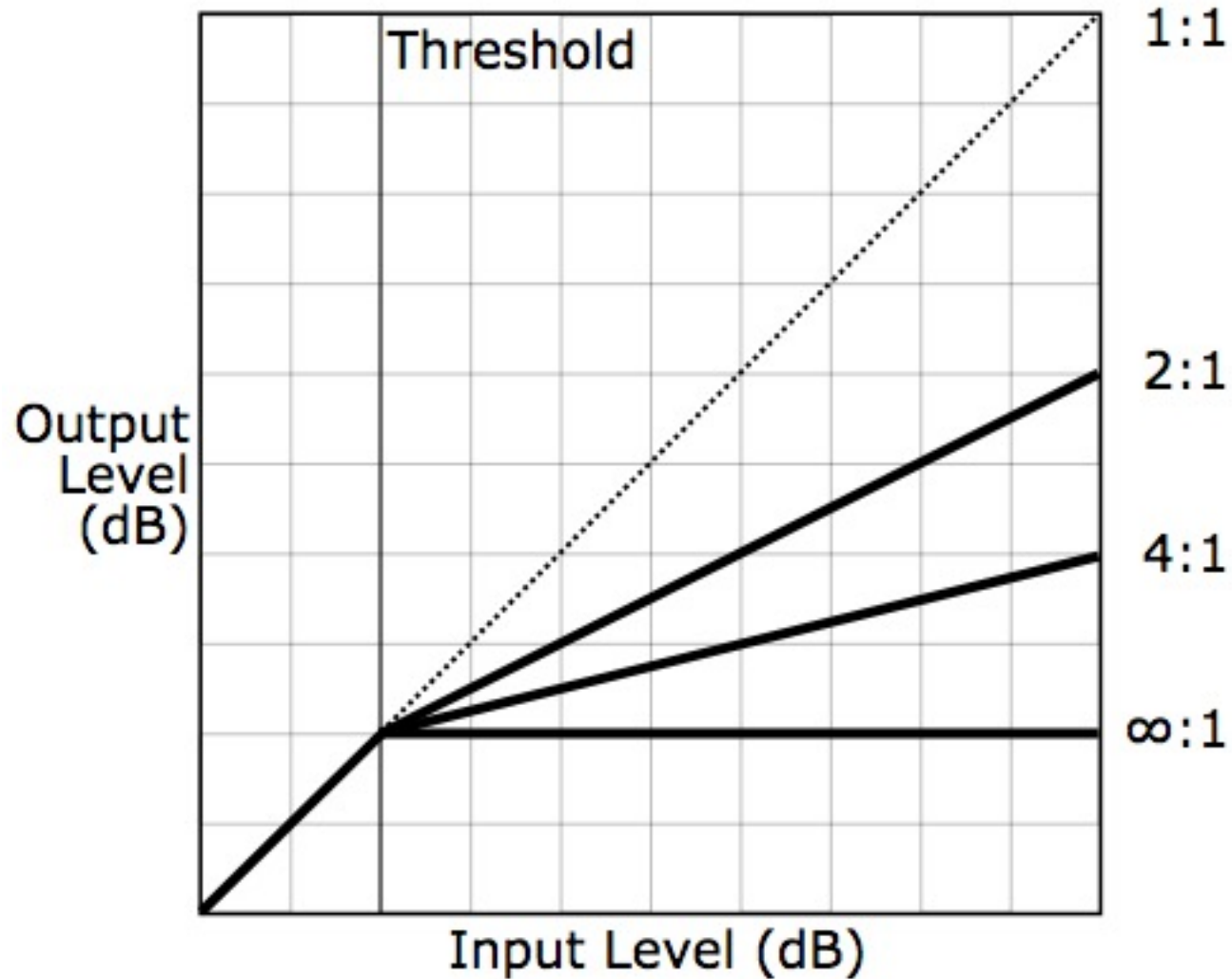
- How fast compression stops after the input signal goes below the threshold



Other compressor settings

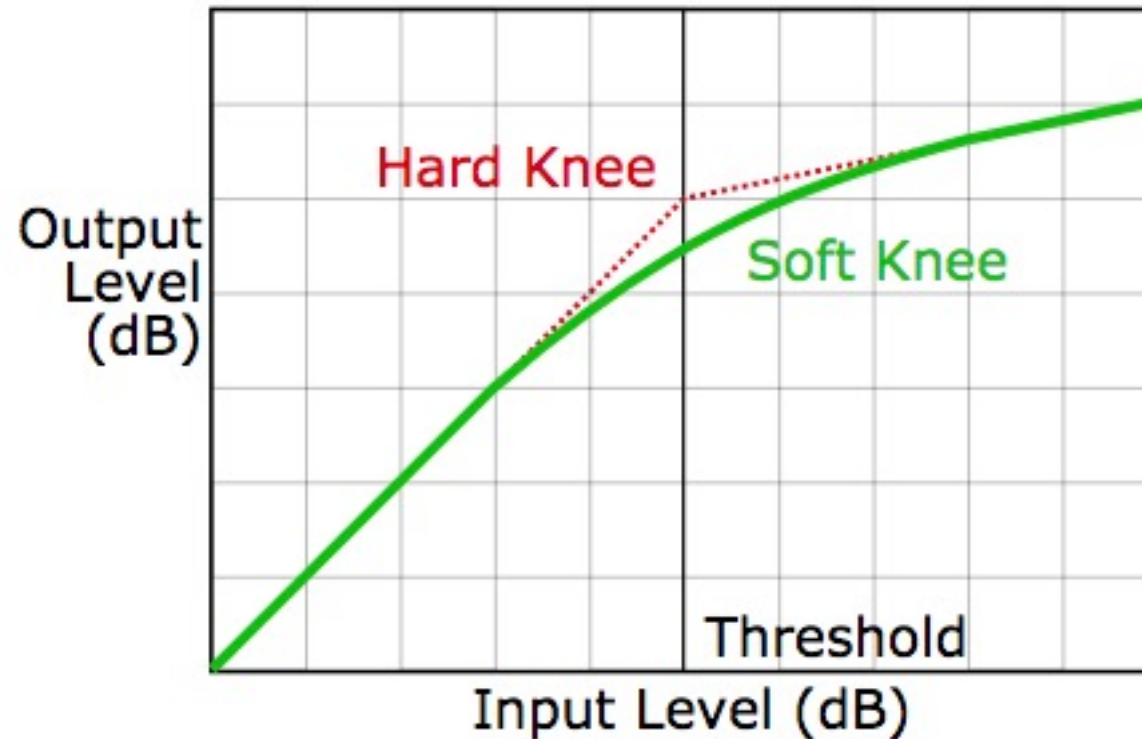
- **Makeup Gain:** Compression reduces the overall loudness of the input signal. If that is not desired, the output signal can be gained to compensate for it
 - Some compressor plugins have automatic makeup gain
- **Knee:** Whether the transition over the threshold is abrupt (hard knee) or gradual (soft knee)
- **Noise floor:** Parts of the signal below the noise floor will not be gained up
- **Peak vs RMS sensing:** Whether the peak level or the RMS level of the input signal is compared to the threshold

Compression transfer function



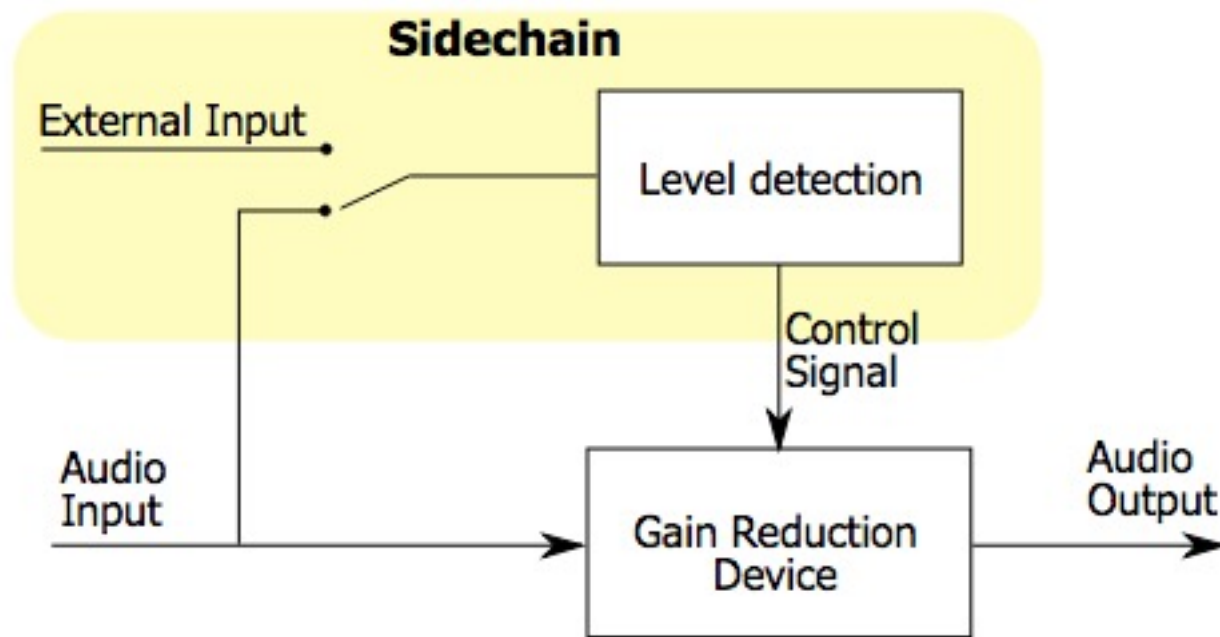
Hard knee vs soft knee

- Soft knee may be desired with high compression ratios to avoid having the compression effect being obvious



Sidechaining

- Instead of compressing when the input signal is above the threshold, a compressor can be set to compress when an external signal is above the threshold
- Used to lower the music, when the DJ speaks



Compression uses

- Even out the volume of a vocal track if the singer has been moving with regard to the microphone
- Increase the sustain of a musical instrument
 - By softening the onset of a note, the decay lasts longer
- Reduce the dynamic range of instruments with wide variation in loudness, such as drums
- Compressing each instrument ensures that they will all be heard in the final mix

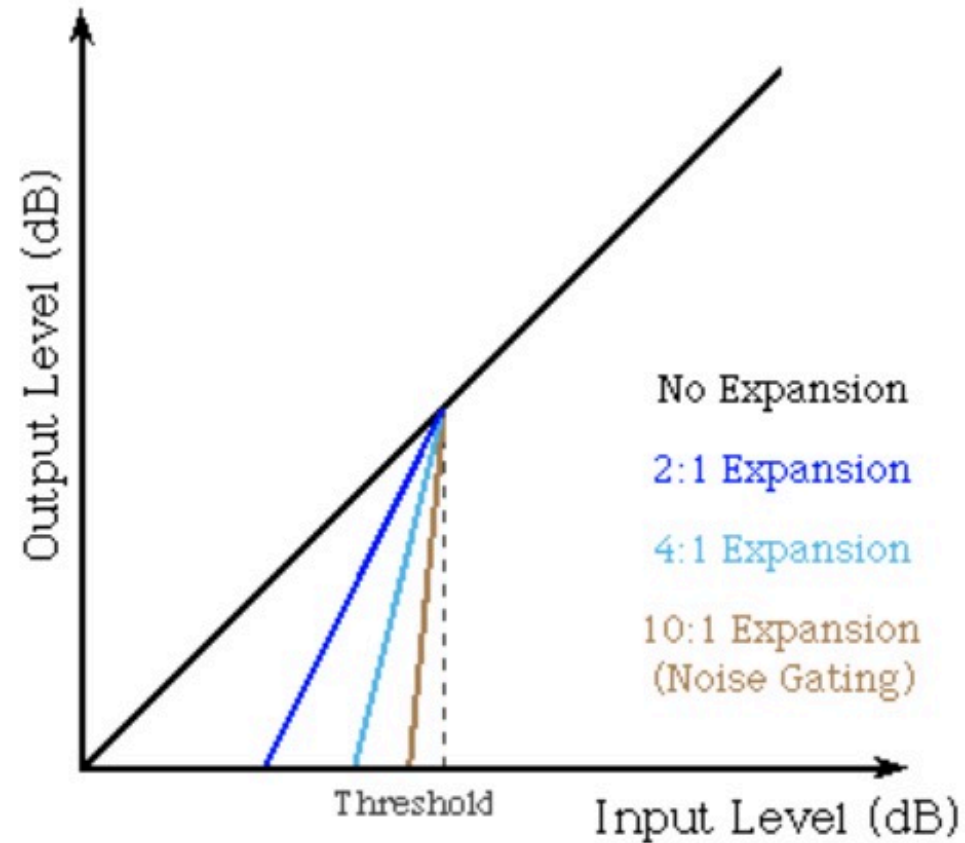
Limiting

- A limiter is a compressor with a very high ratio (10:1 or more) and a fast attack time
- Brickwall limiters (ratios of 20:1 and higher) ensures audio is never above the threshold
 - Used as safety device in live sound and broadcast applications

Dynamic range expansion

- The dual of a compressor is an expander
- An expander makes the softer parts of a signal even softer
- Used to soften unwanted sounds, such as bleed from other instruments, or background noise
- Has similar settings to a compressor
 - Threshold, expansion ration, attack, release

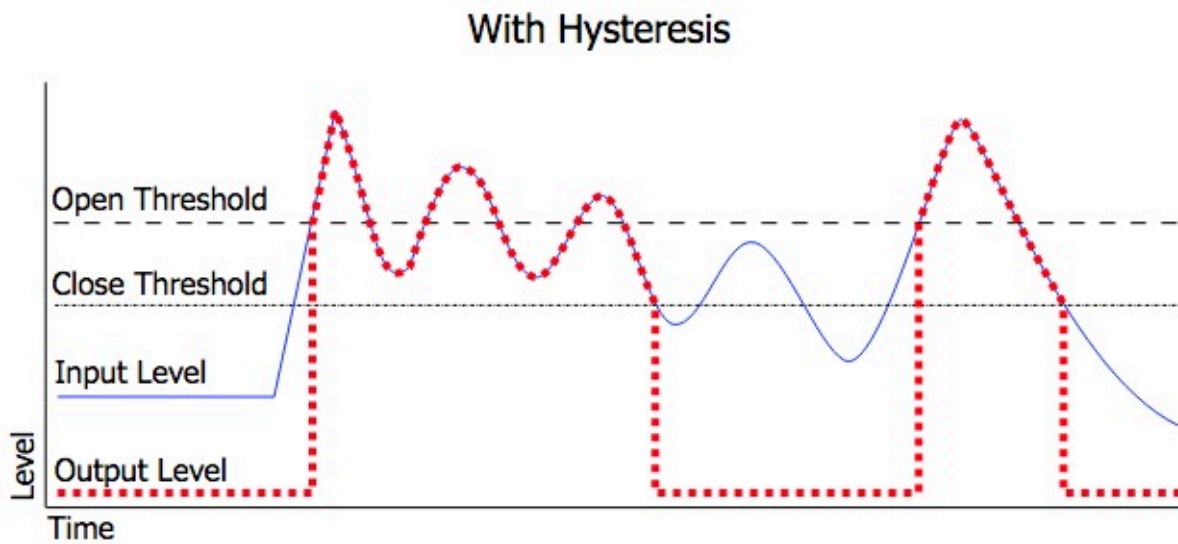
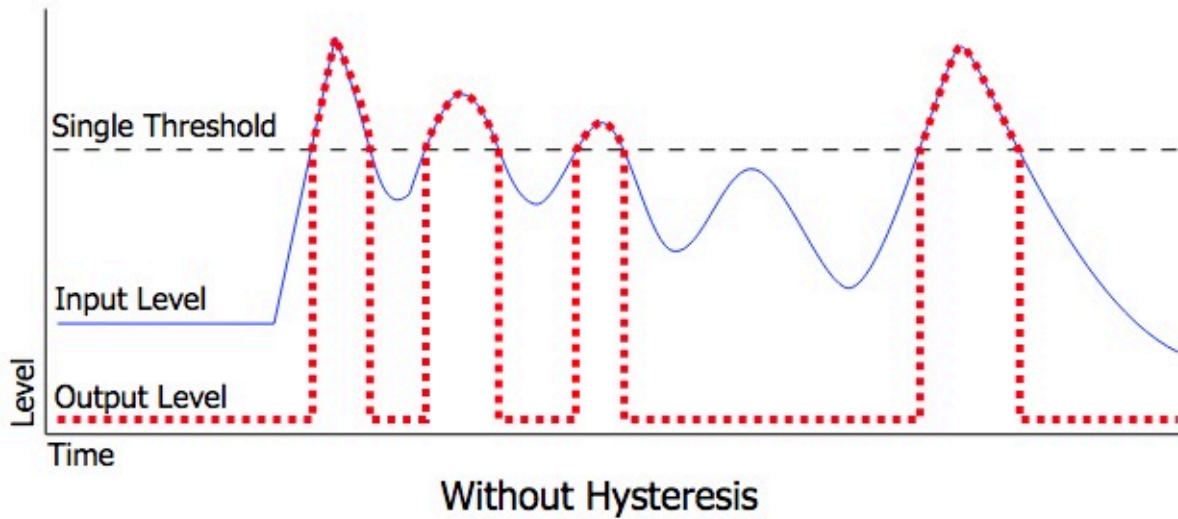
Expansion transfer function



Noise gates

- Expanders with a large expansion ratio are called noise gates, as no output is produced when the input signal is below the threshold
- Noise gates have an additional parameter: **Hold**
 - Specifies an amount of time that the gate will stay open even if the input signal is below the threshold
 - Used to avoid opening the gate on and off too often
- Some noise gates will have different open and close thresholds to achieve the same goal of not opening and closing the gate too often
 - This feature is called **hysteresis**

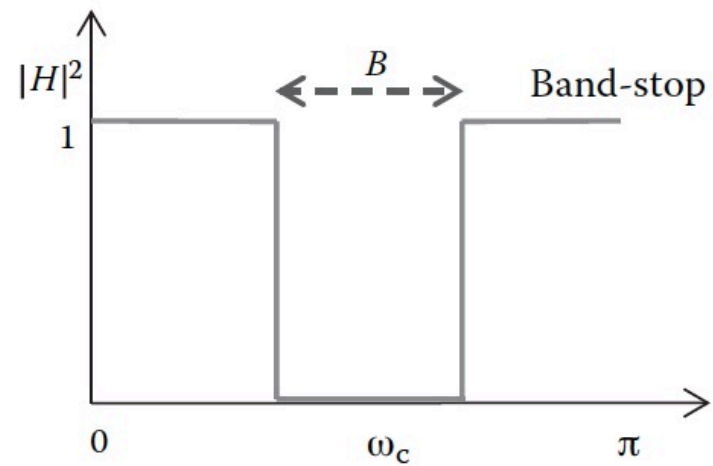
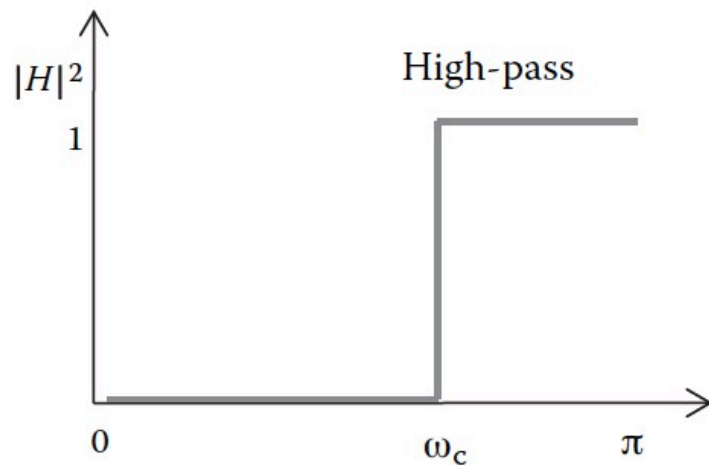
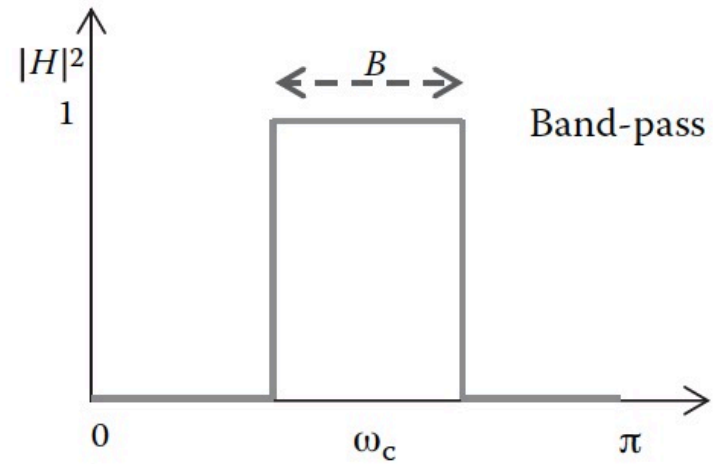
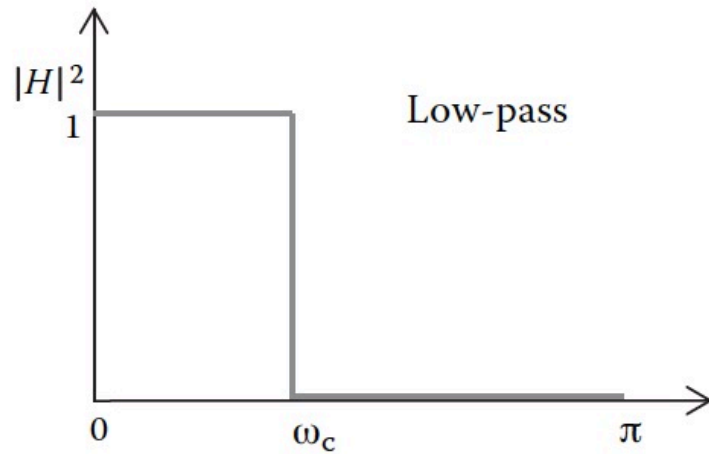
Hysteresis



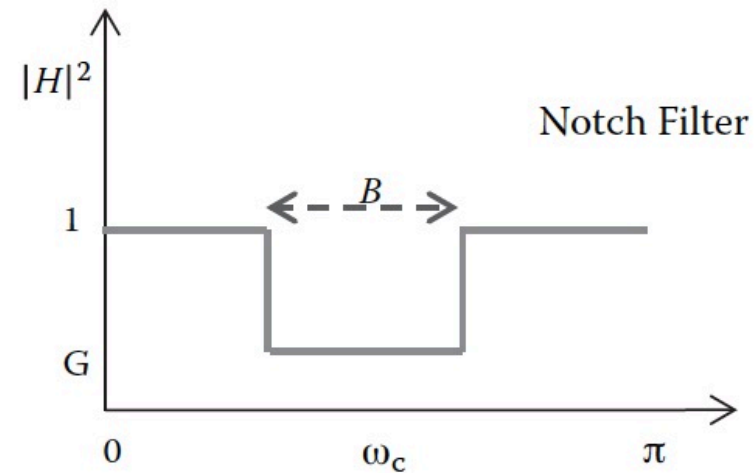
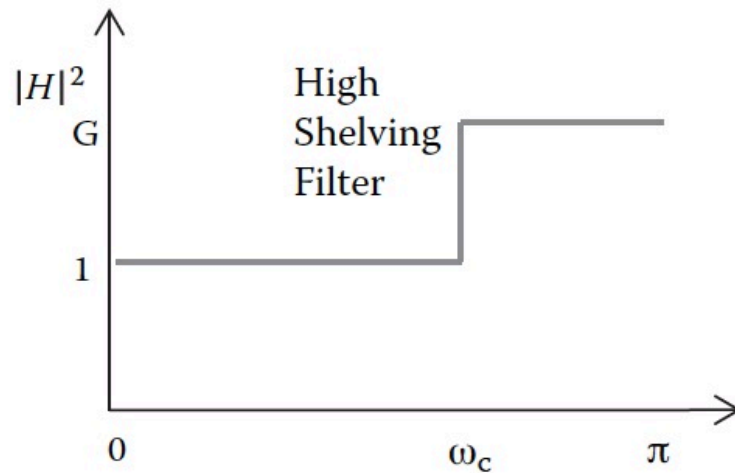
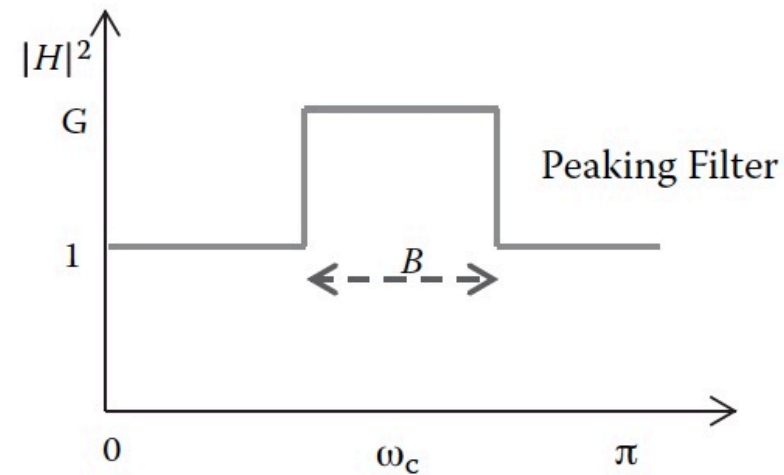
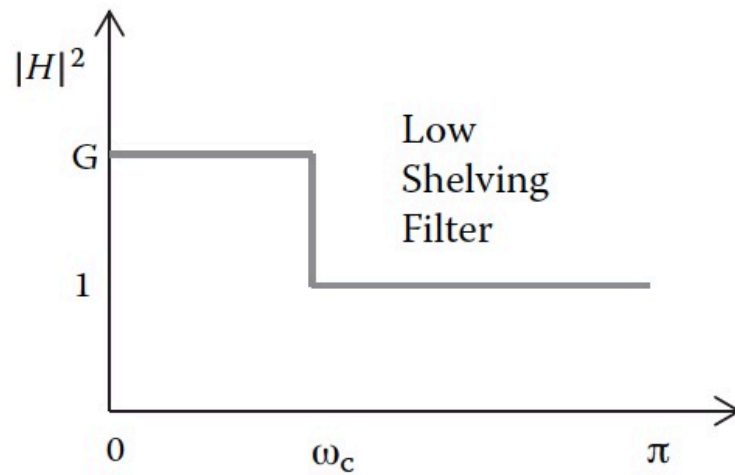
Filters

- Frequency-based effects that affect the distribution of magnitude to different frequencies
- Some filters aim to completely eliminate some frequencies
 - High-pass, Low-pass, Band-pass, Band-stop
- Others adjust the relative gain of different frequencies
 - Shelving, peaking, and notch filters

Ideal transfer functions 1

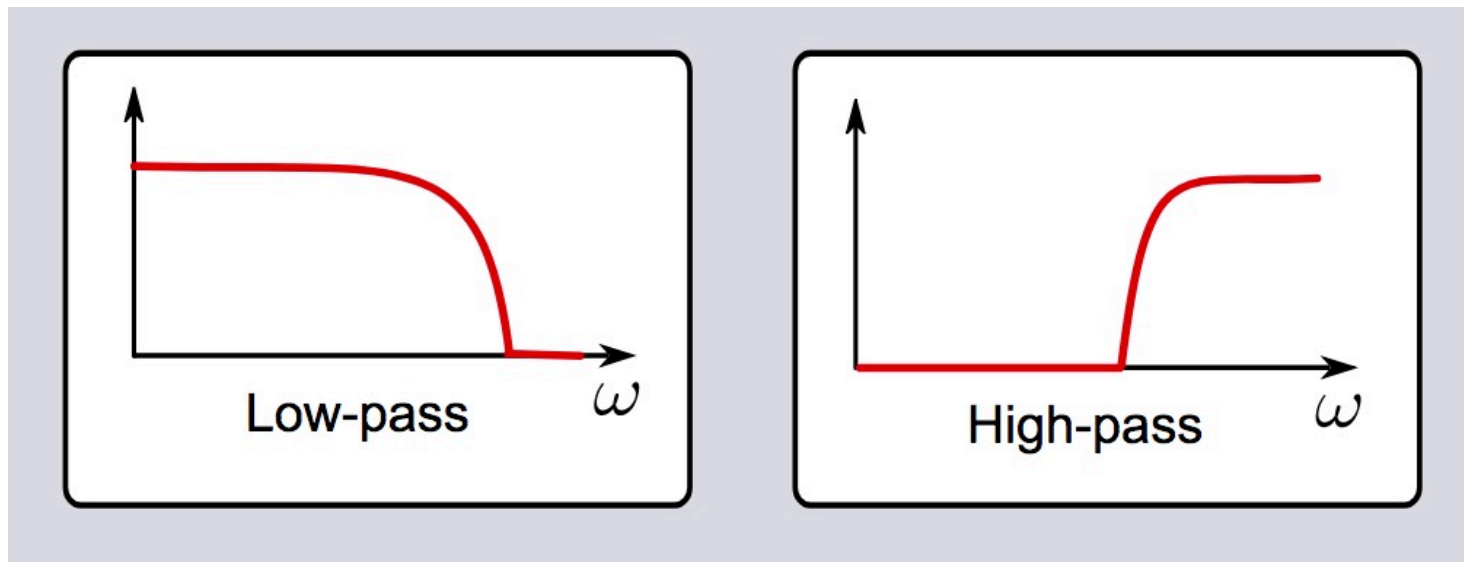


Ideal transfer functions 2



Practical filters

- In practice, it is neither possible nor desirable to have such sharp cutoffs

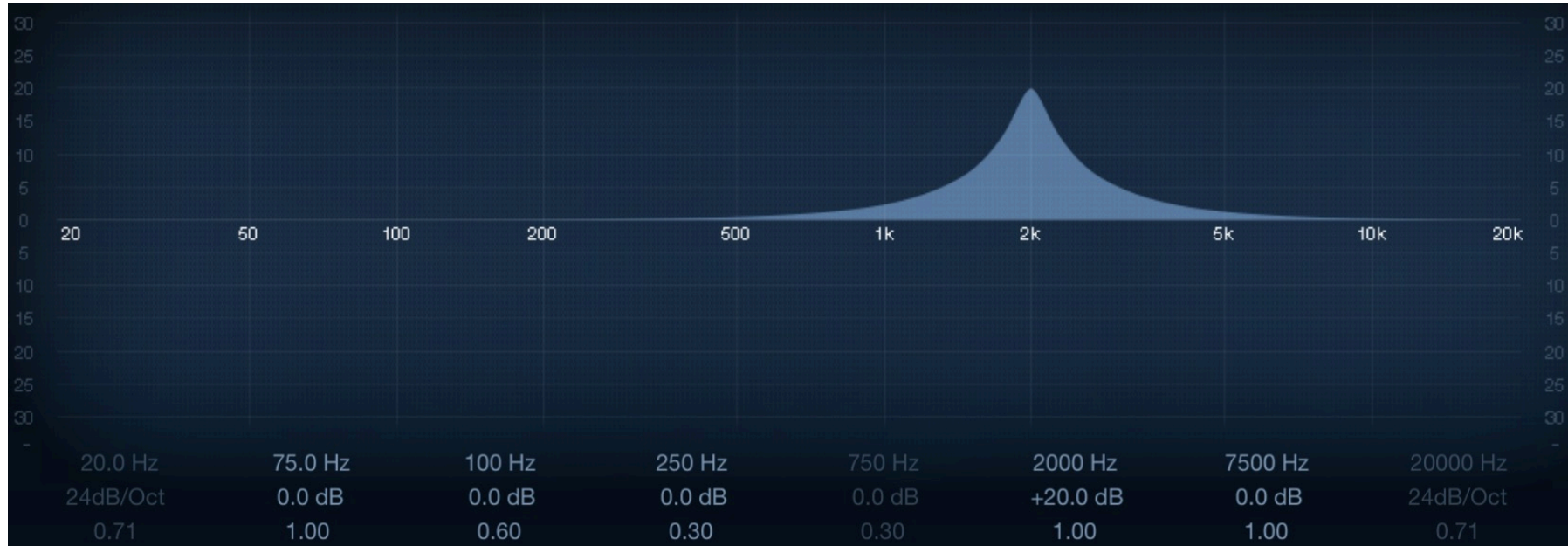


- The cutoff frequency is where attenuation has reached 3dB
- The rolloff factor indicates how fast attenuation happens after the cutoff frequency
 - Often measured in dB per octave

Equalization

- Equalizers are used to balance the relative gain of different frequency bands
- An equalization operation has three parameters
 1. **Center frequency:** The frequency of largest boost or attenuation
 2. **Gain:** The amount of gain for the center frequency
 3. **Bandwidth:** The total range of frequencies affected measured in octaves
 - In DAWs, bandwidth is often shown as Q
 - The smaller the Q , the wider the bandwidth

Equalization operation



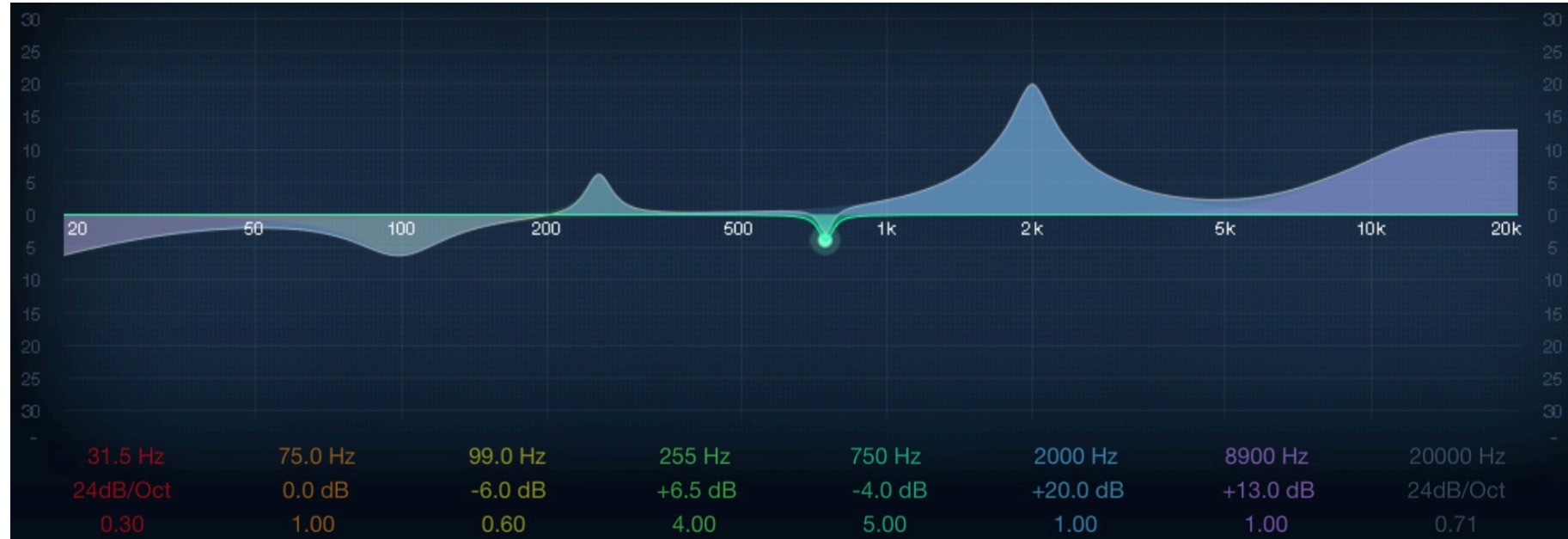
2000 Hz
+20.0 dB
1.00

Center Frequency

Gain

Q (Bandwidth)

EQ Curve



Several filters may be combined to achieve the desired effect

Their combined effect provides the EQ curve of an equalizer effect

Wah-wah

- Essentially a band-pass or peaking filter whose center frequency is changed by a foot pedal
- In typical wah-wahs, the center frequency ranges from 300Hz to 1200Hz
- This simulates formants found in speech
 - The [u] sound has formants around 300Hz
 - The [a] sound has formants at 750Hz and 1200Hz
- **Formants** are stronger frequencies when we speak depending on how the vocal tract is shaped
 - They are the way we recognize different vowels

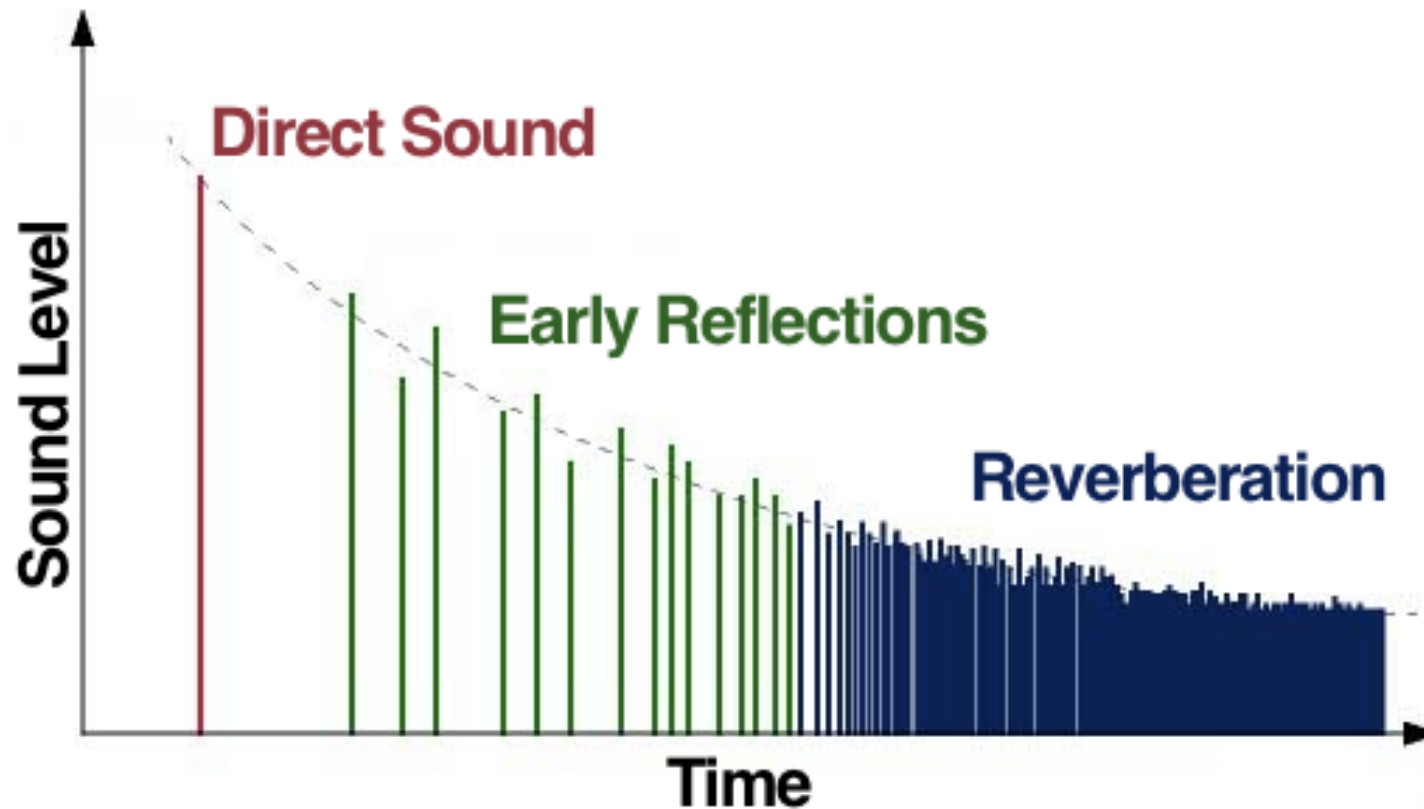
Wah variations

- **Auto wah:** Uses a low-frequency oscillator (**LFO**) typically at around 1-2Hz to adjust the center frequency
 - Some variations adjust the center frequency based on the amplitude of the input signal
- **Tremolo wah:** An auto wah with a second LFO to modify the amplitude of the output signal
 - The two LFO can be independent, synchronized, or even synchronized but out of phase

Reverberation

- Sound in a room reaches the listener in many ways:
 - Directly
 - After one reflection in any of the room surfaces
 - After many reflections in many different surfaces
- Reflected copies of the original sound are attenuated because they travel longer and they are partially absorbed by the surfaces
- As these reflections typically within 40ms of the direct sound, they are perceived as one sound
- The combination of all the copies gives us an impression of space

Room impulse response



Each vertical line is a copy of the original sound

The height of each of line is the amplitude of the copy

Reverberation time

- To summarize the reverberation properties of a room, we use the notion of reverberation time RT_{60}
 - The time it takes for the sound to decay by 60dB
 - Small rooms have RT_{60} that is usually less than 1 sec
 - Larger rooms may have an RT_{60} of 2 sec or more
- Several factors affect the RT_{60} of a room
 - Room dimensions
 - Wall materials
 - Furniture or people in the room