Equalizer

Equalization is a technique for processing audio signals and represents the filtering process to vary the sound content of an audio signal. An equalizer or EQ is a filter as an electronic or digital component for sound shaping and equalization of sound frequencies, mainly music, or of other frequency-based signals such as modulated data signals. Equalization may be used to eliminate or reduce unwanted sounds (e.g., low hum coming from a guitar amplifier), make certain instruments or voices more (or less) prominent, enhance particular aspects of an instrument's tone, or combat feedback (howling) in a public address system.

An equalizer consists of several filters with which the spectrum of the input signal can be processed. Usually the EQ is used to correct linear distortion of a signal. It can be differentiated between different design and operating concepts.

Usage

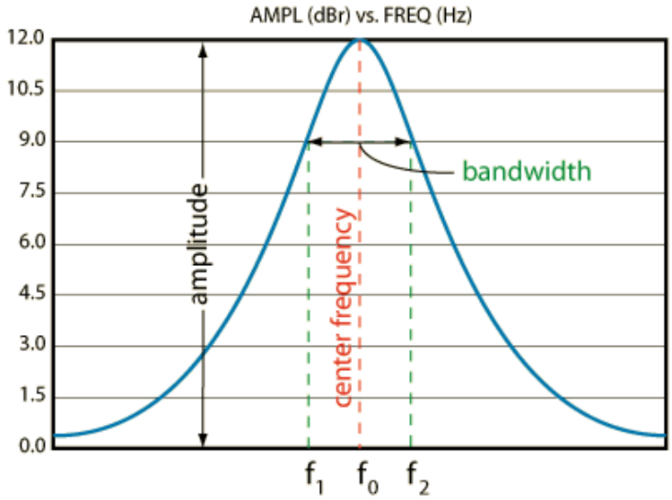
The fields of application of equalization are manifold:

- timbre correction during sound recording
- correction during the post-production phase of the timbre of recorded signals
- elimination / reduction of rustling, recurring noises
- creation of new, manipulated or simulated sounds (for example: Simulation of a phone call in a show)
- Adapt the overall sound of the sound system ("PA system") to the location
- DJs also use equalizers, which are integrated into the DJ mixing consoles (also called mixers)

The Parameters

There are three elementary parameters: center frequency, gain/amplitude, filter quality Q

Center Frequency: This is the frequency that is most boosted or cut by the equalizer. Not only is the set frequency changed for itself, but also the ones around it - this is the frequency at which the maximum gain (or minimum) of the bell is reached. The width of the change is influenced by the slope or filter quality Q.



Gain: Is given in decibels (dB) and stands for the increase or decrease in volume (change in amplitude) at the set frequency that results from processing with the EQ.

Filter quality (Q factor): In equalizers, Q is the ratio of center frequency to bandwidth. It shows the steepness of the filter curve and thus the area around the center frequency to be manipulated - Q denotes the slope of the used filter. It is a parameter that measures the amplitude of the bell, means the amplitude of the band of frequencies that are amplified (or attenuated). How large the area around the selected center frequency (frequency band) is to be processed by the equalizer can be set with Q.

Types of Equalizer

Graphic equalizer

Each frequency band that can be influenced is assigned its own controller (as an independent device it has 26 to 33, typically 31 frequency bands each 1/3 octave wide), so that the course of the frequency correction is “graphically” represented by the controller. This type of equalizer is also called an x-band equalizer. The x stands for the number of controls (with 31 controls as a 31-band equalizer). Smaller graphic equalizers (approx. 2–10 channels) are used as tone controls in some power amplifiers.

Parametric equalizer

Parametric Equalizers are more complex but allow you to make corrections avoiding particular damage to the signal. Being used particularly in filming sounds in live shows, where errors and heavy tonal corrections can be harmful.

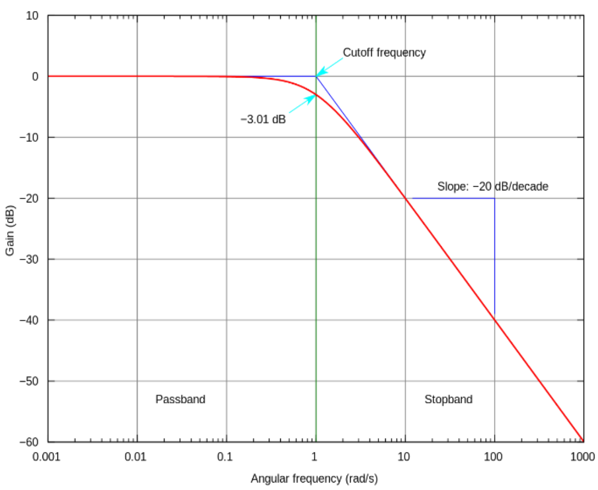
A parametric equalizer is on average equipped in analog mixers with 4 filters instead of 7, 15 or even 31 as in the graphic types, but each of these filters can be adjusted in its parameters, hence the name. Parametric EQs allows to act more targeted than graphical EQs, which instead have a simpler interface.

The center frequency and the change in amplitude (semi-parametric equalizer) as well as often the filter quality Q (corresponding to the bandwidth) can be set here (fully parametric equalizer) for one or more frequency bands. This type of construction can be found mainly in mixing consoles, effects devices and as equalization plugins.

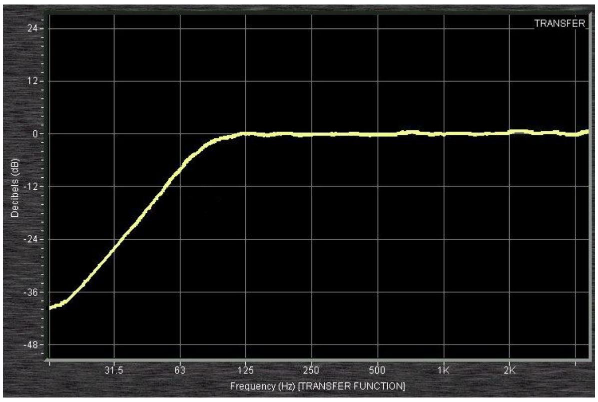
EQ Filter

High Pass (HPF) and Low Pass (LPF) Filters

A **low-pass filter** weakens the high frequencies up to a cut-off frequency and allows all lower frequencies to pass with practically no attenuation. This "low pass" is e.g. used as an anti-aliasing filter (see aliasing effect) or for noise reduction. Low-pass filters also play a significant role in the sculpting of sound created by analogue and virtual analogue synthesisers.



A **high-pass filter** weakens the low frequencies to a cut-off frequency, while all higher frequencies are passed.

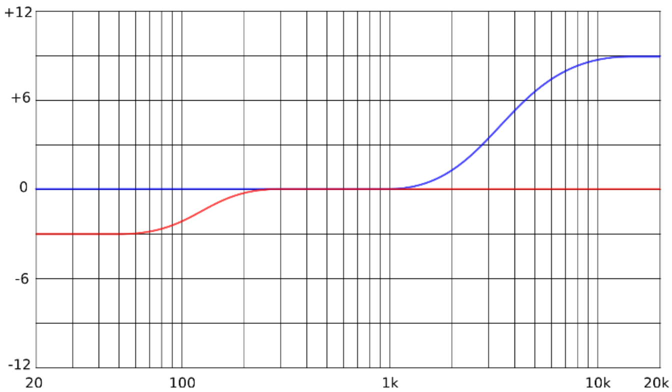


The two parameters of these filters are frequency and slope. The attenuation of what is cut depends on the slope of the filter, which can usually be 6, 12, 18 or 24 dB per octave. In the analog world we usually have only the frequency selection but in digital we often also have the possibility to choose the slope and sometimes the emphasis at the cutting point (resonance).

Shelving filters

While high and low pass filters are useful for removing unwanted signals above or below a set frequency, shelving filters can be used to reduce or increase signals above or below a set frequency.

They are used for high and low frequencies and emphasize or attenuate the selected frequency downwards (low shelving) or upwards (high shelving). They therefore usually have two parameters, selected frequency and dB gain / attenuation. Sometimes there is a third parameter that controls the slope at the gain or damping point.



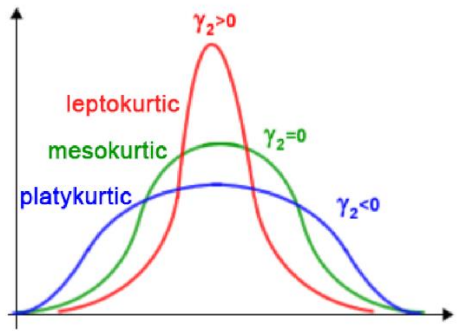
Control curves of two shelving filters (Blue Line: treble boost; red line: bass cut)

Shelving filters are used as common tone controls (bass and treble) found in consumer audio equipment such as home stereos, and on guitar amplifiers and bass amplifiers.

For a brief simplified description how to use a EQ: https://www.musicianonamission.com/approach-equalization-two-types-eq/

Parametric Filters or Peaking Filter

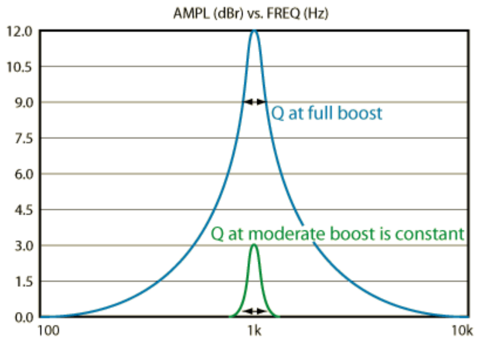
In contrast to shelving filters, the Peak EQ - also called bell filter - can work much more precisely. This EQ type got its name from the peak (peak) or the indentation in bell shape (Bell). Peak filters process a relatively precisely definable frequency range. How much the frequencies around the selected one will be attenuated or emphasized will depend on the parameter of bandwidth or Q: The higher the value of Q the narrower the bell of gain and attenuation will be. The three parameters in this case will be frequency, gain / attenuation and bell width. If one of these three parameters are missing, the filter is called semi-parametric.



High Q = very pointed curve (leptokurtic)

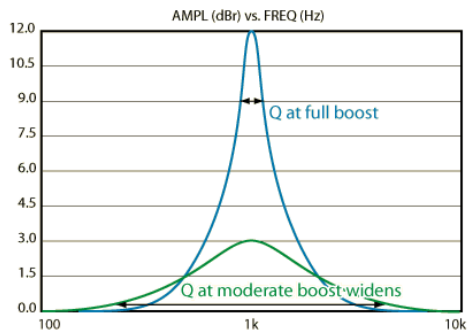
Low Q = very flat curve (platykurtic)

Particular peaking filters



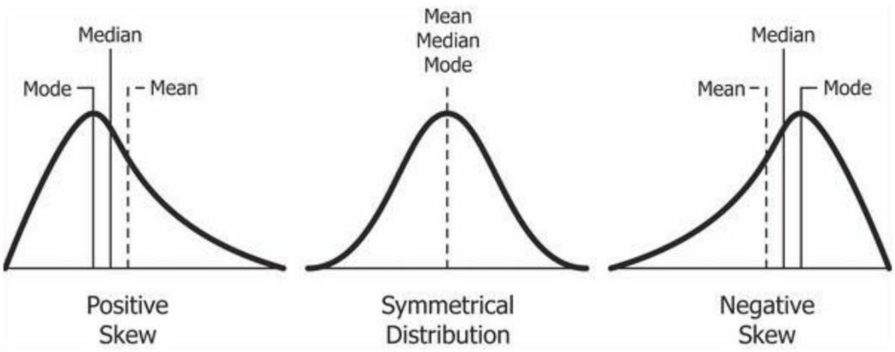
Constant Q (semiparametric)

Q remains constant with changes in dB



Proportional Q (semiparametric)

Q varies proportionally to the increase or decrease in dB



Bells not

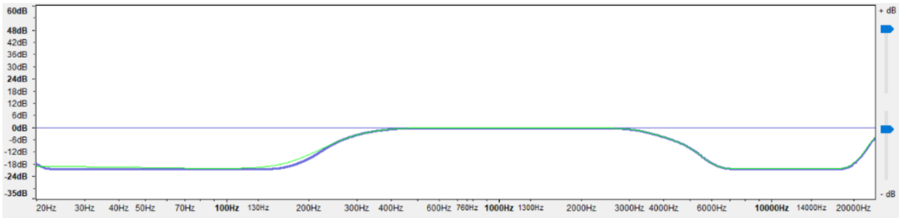
symmetrical

Some parametric filters have a fixed Q value with varying gain or attenuation (Constant Q), others have variable bandwidth (Proportional Q) which tightens with increasing gain or attenuation. This in order to increase the precision in case of cuts or gains on specific frequencies.

Certain filters have non-symmetrical bells in the event of cutting or damping, such as the peaking filter of Waves' Renaissance Equalizer. The bell tightens when cut, to remove precise frequencies and it widens in case of gain so as not to be too resonant on a specific frequency.

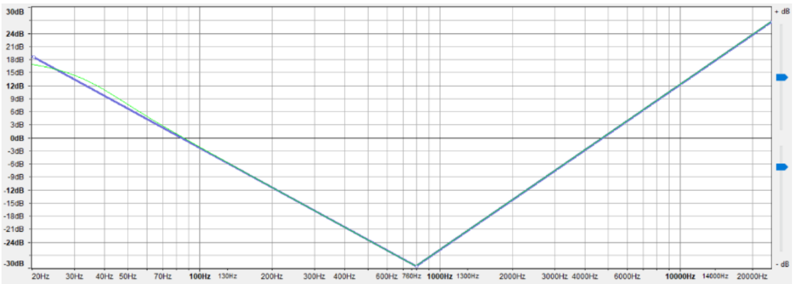
Examples of filters that deliver specific sounds

Phone Filters:



Both high and low frequencies are damped. Essentially, all frequencies except the 300Hz to 4000Hz range are damped**.**

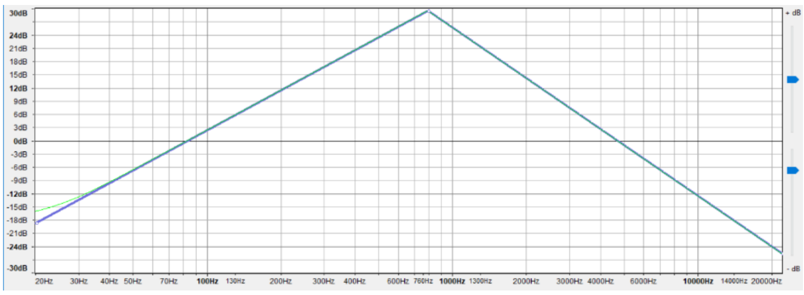
Midcut filter:



The midcut or "V" filter amplifies low and high frequencies. It does not cancel the mid frequencies but attenuates them while some cancel them.

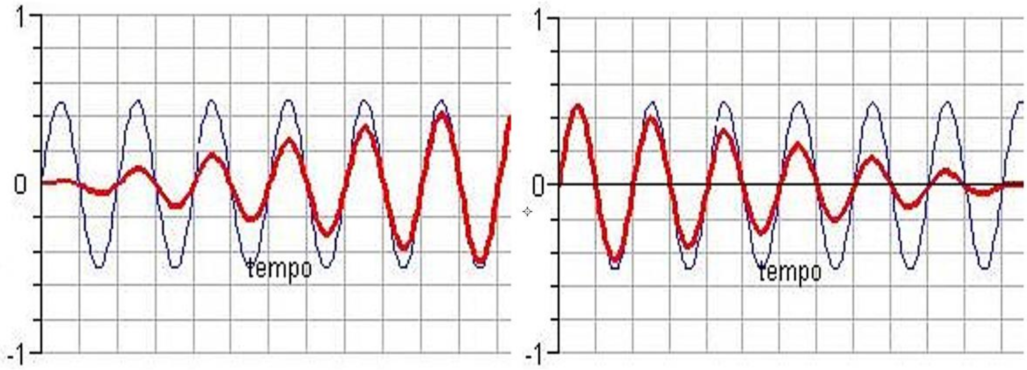
Applying this filter produces a somewhat empty and metallic sound.

Medium-pass filter:



The mirror image of the medium cut dampens the low and high frequencies and amplifies the middle ones. Applying this filter creates a nasal sounding effect.

Fade-In and Fade-Out

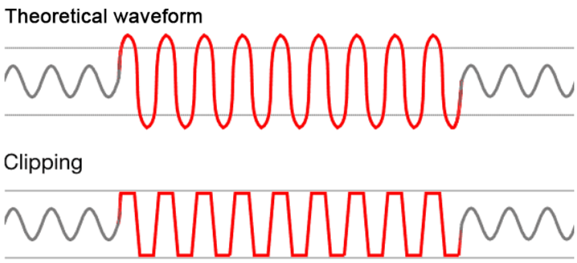


Fade-in and fade-out phenomena are usually present in an audio track. They are the gradual increase and the gradual decrease in an audio signal. This can be used anywhere and is not limited to the beginning and end of a recording.

Clipping

In signal processing, overdriving is the act on signal processing units with input signals outside the permitted input range. Undesired effects occur as a result of an overload. In digital signal processing, clipping occurs when the signal is restricted by the area of a selected display. Typically, excursions that go beyond the input area are cut off. This effect is known as clipping.

It is theoretically possible that while the sound is being processed with the EQ, frequencies are amplified which later lie outside the displayable range or the performance range of the audio playback devices (see previous lectures).



It can also happen that an amplifier generates an output signal that is higher than the supply voltage. The signal is only amplified to the maximum capacity, and at this point the signal cannot be amplified further.

Depending on the frequency and length of the clipped waveform areas, the sound result of "clipping" varies. In subtle form, it is practically inaudible, but with increasing intensity, very harsh, scratchy, sharp-edged sounding distortions are caused. Hard clipping is sometimes used specifically - to maximize loudness during mastering or for the harsh distortion mentioned. Strong hard clipping can damage or ruin speakers (especially their tweeters).