Equalization (audio)

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(Redirected from Graphic equalizer)

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A stereo graphic equalizer

Notes	2/3-octave, 15 bands per channel

Equalization is the process commonly used in sound recording and reproduction to alter the frequency response of an audio system using linear filters. Most hi-fi equipment uses relatively simple filters to make bass and treble adjustments. Graphic and parametric equalizers have much more flexibility in tailoring the frequency content of an audio signal. An **equalizer** is the circuit or equipment used to achieve equalization.[1][2] Since equalizers, "adjust the amplitude of audio signals at particular frequencies," they are, "in other words, frequency-specific volume knobs."[3]

Equalizers are used in recording studios, broadcast studios, and live sound reinforcement to correct the response of microphones, instrument pick-ups, loudspeakers, and hall acoustics.[2] Equalization may also be used to eliminate unwanted sounds, make certain instruments or voices more prominent, enhance particular aspects of an instrument's tone, or combat feedback (howling) in a public address system.[1][2] Equalizers are also used in music production to adjust the timbre of individual instruments by adjusting their frequency content and to fit individual instruments within the overall frequency spectrum of the mix.[4]

The most common equalizers in music production are parametric, semi-parametric, graphic, peak, and program equalizers.[5] Graphic equalizers are often included in consumer audio equipment and software which plays music on home computers. Parametric equalizers require more expertise than graphic equalizers, and they can provide more specific compensation or alteration around a chosen frequency. This may be used in order to remove (or to create) a resonance, for instance.

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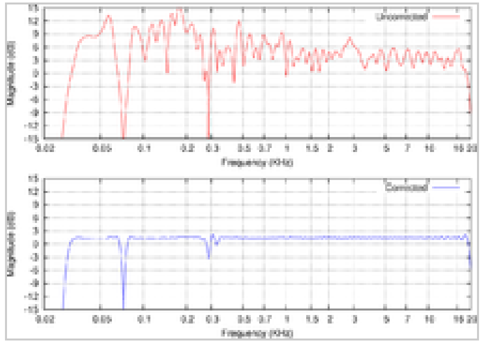
Terminology

The concept of equalization was first applied in correcting the frequency response of telephone lines using passive networks; this was prior to the invention of electronic amplification. Initially equalization was used to "compensate for" (i.e. correct) the uneven frequency response of an electric system by applying a filter having the opposite response, thus restoring the fidelity of the transmission. A plot of the system's net frequency response would be *flat*, as its response to all frequencies would literally be *equal*. Hence the term "equalization."



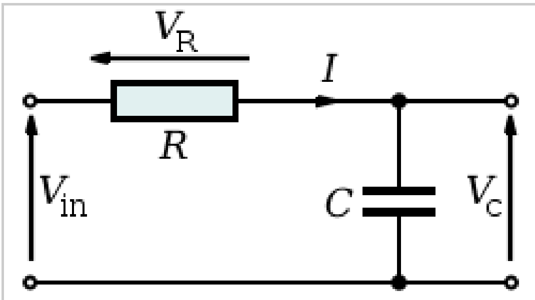
The very uneven spectrum of white noise played through imperfect speakers and modified by room acoustics (top) is *equalized* using a sophisticated filter using digital hardware (bottom). The resulting "flat" response fails, however, at 75 Hz where the original system had a null in its response which cannot be corrected.

Much later the concept was applied in audio engineering to adjust the frequency response in recording, reproduction, and live sound reinforcement systems. Sound engineers correct the frequency response of a sound system so that the frequency balance of the music as heard through speakers better matches the original performance picked up by a microphone. Audio amplifiers have long had filters or controls to modify their frequency response. These are most often in the form of variable bass and treble controls (shelving filters), and switches to apply low-cut or high-cut filters for elimination of low frequency "rumble" and high frequency "hiss" respectively.

Graphic equalizers and other equipment developed for improving fidelity have since been used by recording engineers to modify frequency responses for aesthetic reasons. Hence in the field of audio electronics the term "equalization" is now broadly used to describe the application of such filters regardless of intent. This broad definition therefore includes all linear filters at the disposal of a listener or engineer.

Filter functions

The responses of linear filters are mathematically described in terms of their transfer function or, in layman's terms, frequency response. A transfer function can be decomposed as a combination of *first order* responses and *second order* responses (implemented as so-called biquad sections). These can be described according to their so-called pole and zero frequencies, which are complex numbers in the case of second-order responses.



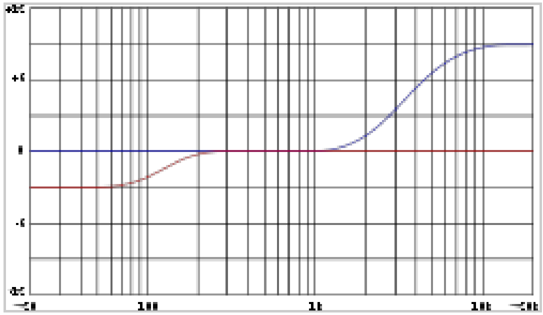
A first order low-pass (high-cut) filter implemented using only a resistor and capacitor.

First order filters

A first order filter can alter the response of frequencies above and below a point. In the transition region the filter response will have a slope of up to 6 dB per octave. The bass and treble controls in a hi-fi system are each a first order filter in which the balance of frequencies above and below a point are varied using a single knob. A special case of first order filters is a first order high-pass or low-pass filter in which the 6 dB per octave cut of low or high frequencies extends indefinitely. These are the simplest of all filters to implement individually, requiring only a capacitor and resistor.

Second order filters

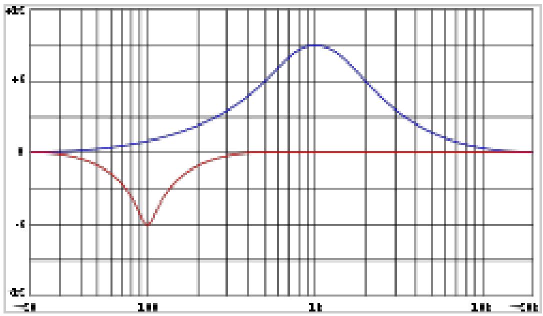
Second order filters are capable of resonance (or anti-resonance) around a particular frequency. The response of a second order filter is specified not only by its frequency but its Q; a higher Q corresponds to a sharper response (smaller bandwidth) around a particular center frequency. For instance, the red response in the accompanying image cuts frequencies around 100 Hz with a higher Q than the blue response which boosts frequencies around 1000 Hz. Higher Q's correspond to resonant behaviour in which the half-power or -3 dB bandwidth, *BW*, is given by:



Two first-order shelving filters: a -3dB bass cut (red), and a +9dB treble boost (blue)

$$BW = F_0/Q$$

where F_0 is the resonant frequency of the second order filter. *BW* is the bandwidth expressed in the same frequency unit that F_0 is. Low Q filter responses (where $Q < \frac{1}{2}$) are not said to be resonant and the above formula for bandwidth does not apply.



Second order filter responses

It is also possible to define the Q of a band-pass function as:

$$Q = \frac{\sqrt{2^N}}{2^N - 1} = \frac{1}{2\sinh\left(\frac{\ln(2)}{2}N\right)},$$

where *N* is the bandwidth in octaves. It should be noted that a second-order filter response with Q of less than 1/2 can be decomposed into two first-order filter functions, a low-cut and a high-cut (or boost). Of more interest are resonant filter functions which can boost (or cut) a narrow range of frequencies. In addition to specifying the center frequency F_0 and the Q, the specification of the filter's zeros determines how much that frequency band will be boosted (or cut). Thus a parametric equalizer section will have three controls for its center frequency F_0, bandwidth or Q, and the amount of boost or cut usually expressed in dB.

The range of second-order filter functions is important because any analog filter function can be decomposed into a (usually small) number of these (plus, perhaps, simpler first order responses). These are implemented directly by each section of a parametric equalizer where they are explicitly adjusted. And each element of a graphic equalizer based on a filter bank includes one such element whose Q is not adjustable by the user.

Filter types

High and low pass filters

A high-pass filter is a filter that passes higher frequencies well but attenuates lower frequency components. A low-pass filter passes low-frequency components of signals while attenuating higher frequencies. In audio applications these are frequently termed "low cut" and "high cut" respectively, to emphasize their effect on the original signal. For instance, sometimes audio equipment will include a switch labeled "high cut" or described as a "hiss filter" (hiss being high frequency noise). In the times of phonographs many stereos would include a switch to introduce a high-pass (low cut) filter, often called a "rumble filter," to eliminated infrasonic frequencies.

Shelving filter

Common tone controls (bass and treble) found in consumer audio equipment are examples of variable *shelving* *filters*. These implement a first order response, as discussed above and provide an adjustable boost or cut to frequencies above or lower than a certain point. A *high shelf* or "treble control" will have a frequency response |*H(f)*| whose square is given by:

$$|H(f)|^2 = \frac{1 + (f/f_z)^2}{1 + (f/f_p)^2}$$

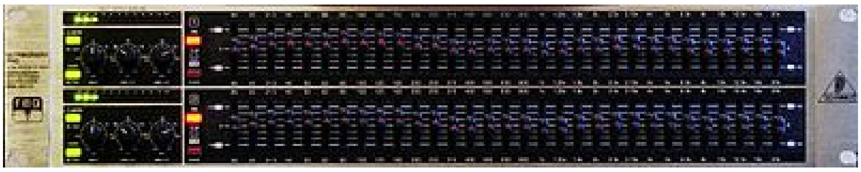
where f_p and f_z are called the pole and zero frequencies, respectively. Turning down the treble control increases f_z so that frequences higher than f_p are attenuated. Turning up the treble control increases f_p so that frequences higher than f_z are boosted. Setting the treble control at the center sets $f_z = f_p$ so that $|H(f)|^2=1$ and the circuit has no effect. At most, the slope of the filter response in the transition region will be 6 dB per octave (thus a doubling of signal voltage and a consequent quadrupling of signal power for every doubling of frequency).

Similarly the response of a *low shelf* or "bass control" can be represented as

$$|H(f)|^2 = (f_z/f_p)^2 \, \frac{1 + (f/f_z)^2}{1 + (f/f_p)^2}.$$

In this case the inclusion of the leading factor simply indicates that the response at frequencies much higher than f_z or f_p is unity and that only bass frequencies are affected. Note that a high shelve in which f_z is set to infinity or a low shelve response in which f_z is set to zero implements a first order low-pass or high-pass filter respectively. However usual tone controls have a more limited range, since the purpose isn't to eliminate any frequencies but only to achieve a greater balance when, for instance, the treble is lacking and the sound isn't crisp. Since the range of possible responses from shelving filters is so limited, they are considered quite inadequate for equalization tasks among audio engineers.

Graphic equalizer



In the *graphic equalizer*, the input signal is sent to a bank of filters. Each filter passes the portion of the signal present in its own frequency range or *band*. The amplitude passed by each filter is adjusted using a slide control to boost or cut frequency components passed by that filter. The vertical position of each slider thus indicates the gain applied at that frequency band, so that the knobs resemble a *graph* of the equalizer's response plotted versus frequency.

The number of frequency channels (and therefore each one's bandwidth) affects the cost of production and may be matched to the requirements of the intended application. A car audio equalizer might have one set of controls applying the same gain to both stereo channels for convenience, with a total of five to ten frequency bands. On the other hand, an equalizer for professional live sound reinforcement typically has some 25 to 31 bands, for more precise control of feedback problems and equalization of room modes. Such an equalizer (as shown above) is called a 1/3-octave equalizer (spoken informally as "*third-octave* EQ") because the center frequency of its filters are spaced one third of an octave apart, three filters to an octave. Equalizers with half as many filters per octave are common where less precise control is required—this design is called a 2/3-octave equalizer.

Graphic equalizers are sometimes used by stereo owners to obtain a smiley face curve (also known as "mid scoop")[6] in which the lower and higher frequency channels are boosted relative to the midrange frequencies. This results in a graphical pattern resembling a cartoon "smile."

Parametric equalizer

Parametric equalizers are multi-band variable equalizers which allow users to control the three primary parameters: amplitude, center frequency and bandwidth. The amplitude of each band can be controlled, and the center frequency can be shifted, and bandwidth ("Q") can be widened or narrowed. Parametric equalizers are capable of making much more precise adjustments to sound than other equalizers, and are commonly used in sound recording and live sound reinforcement. Parametric equalizers are also sold as standalone outboard gear units.

A variant of the parametric equalizer is the semi-parametric equalizer, also known as a sweepable filter. It allows users to control the amplitude and frequency, but uses a pre-set bandwidth of the center frequency. In some cases, semi-parametric equalizers allow the user to select between a wide and a narrow preset bandwidth.

The equaliser section from the Audient ASP8024 Mixing console. The upper section has high and low shelving EQ, the lower section has fully parametric EQ.

Uses

In sound recording, equalization is used to improve an instrument's sound or make certain instruments and sounds more prominent. For example, a recording engineer may use an equalizer to make some high-pitches in a vocal part louder while making low-pitches in a drum part quieter.[1][2]

Equalization is commonly used to increase the 'depth' of a mix, creating the impression that some sounds in a mono or stereo mix are farther or closer than others, relatively.[7] Equalization is also commonly used to give tracks with similar frequency components complimentary spectral contours, known as **mirrored equalization**. Select components of parts which would otherwise compete, such as bass guitar and kick drum, are boosted in one part and cut in the other, and vice versus, so that they both stand out.[8]

Equalizers can correct problems posed by a room's acoustics, as an auditorium will generally have an uneven frequency response especially due to standing waves and acoustic dampening. The frequency response of a room may be analyzed using a spectrum analyzer and a pink noise generator for instance. Then a graphic equalizer can be easily adjusted to compensate for the room's acoustics. Such compensation can also be applied to tweak the sound quality of a recording studio in addition to its use in live sound reinforcement systems and even home hi-fi systems.

During live events where signals from microphones are amplified and sent to speaker systems, equalization is not only used to "flatten" the frequency response but may also be useful in eliminating feedback. When the sound produced by the speakers is picked up by a microphone, it is further reamplified; this recirculation of sound can lead to "howling" requiring the sound technician to reduce the gain for that microphone, perhaps sacrificing the contribution of a singer's voice for instance. Even at a slightly reduced gain, the feedback will still cause an unpleasant resonant sound around the frequency at which it would howl. But because the feedback is troublesome at a particular frequency, it is possible to cut the gain only around that frequency while preserving the gain at most other frequencies. This can best be done using a parametric equalizer tuned to that very frequency with its amplitude control sharply reduced. By adjusting the equalizer for a narrow bandwidth (high Q), most other frequency components will not be affected. The extreme case when the signal at the channel's center frequency is completely eliminated is known as a notch filter.

An equalizer can be used to correct or "flatten" the frequency response of speakers rather than designing the speaker itself to be equalized. For instance, the highly regarded Bose 901 speaker system doesn't use separate woofers and tweeters to cover the bass and treble frequencies, but includes 9 full-range drivers more akin to what one would find in a table radio. However this speaker system is sold with an active equalizer designed to correct the poor frequency balance of those drivers. That equalizer must be inserted into the amplifier system so that the amplified signal that is finally sent to the speakers has its response increased at the frequencies where the response of these drivers falls off, producing a high fidelity reproduction regardless.

Tone controls (usually designated "bass" and "treble") are simple shelving filters included in most hi-fi equipment for gross adjustment of the frequency balance. The bass control may be used, for instance, to increase the drum and bass parts at a dance party, or to reduce annoying bass sounds when listening to a person speaking. The treble control might be used to give the percussion a sharper or more "brilliant" sound, or can be used to cut such high frequencies when they have been overemphasized in the program material or simply to accommodate a listener's preference.

A "rumble filter" is a high pass (low cut) filter with a cutoff typically in the 20 to 40 Hz range; this is the low frequency end of human hearing. "Rumble" is a type of low frequency noise produced in record players and turntables, particularly older or low quality models. The rumble filter prevents this noise from being amplified and sent to the loudspeakers. Some cassette decks have a switchable "Subsonic Filter" feature that does the same thing for recordings.

A crossover network is a system of filters designed to direct electrical energy separately to the woofer and tweeter of a 2-way speaker system (and also to the mid-range speaker of a 3-way system). This is most often built into the speaker enclosure and hidden from the user. However in bi-amplification, these filters operate on the low level audio signals, sending the low and high frequency components to separate amplifiers which connect to the woofers and tweeters respectively.

Equalization is used in a reciprocal manner in certain communication channels and recording technologies. The original music is passed through a particular filter to alter its frequency balance, followed by the channel or recording process. At the end of the channel or when the recording is played, a complementary filter is inserted which precisely compensates for the original filter and recovers the original waveform. For instance, FM broadcast uses a pre-emphasis filter to boost the high frequencies before transmission, and every receiver includes a matching de-emphasis filter to restore it. The white noise that is introduced by the radio is then also de-emphasized at the higher frequencies (where it is most noticeable) along with the pre-emphasized program, making the noise less audible. Tape recorders used the same trick to reduce "tape hiss" while maintaining fidelity. On the other hand, in the production of vinyl records, a filter is used to reduce the amplitude of low frequencies which otherwise produce large amplitudes on the tracks of a record. Then the groove can take up less physical space, fitting more music on the record. The preamp attached to the phono cartridge has a complementary filter boosting those low frequencies following the standard RIAA equalization curve.

History

Audio electronic equipment evolved to incorporate filtering elements as consoles in radio stations began to be used for recording as much as broadcast. Early filters included basic bass and treble controls featuring fix frequency centers, and fixed levels of cut or boost. These filters worked over broad frequency ranges.

The Langevin Model EQ-251A was the first equalizer to use slide controls. It featured two passive equalization sections, a bass shelving filter, and a pass band filter. Each filter had switchable frequencies and used a 15-position slide switch to adjust cut or boost.[9] The first true graphic equalizer was the type 7080 developed by Art Davis's Cinema Engineering. It featured 6 bands with a boost or cut range of 8 dB. It used a slide switch to adjust each band in 1 dB steps. Davis's second graphic equalizer was the Altec Lansing Model 9062A EQ. In 1967 Davis developed the first 1/3 octave variable notch filter set, the Altec-Lansing "Acousta-Voice" system.[10]

Daniel N. Flickinger introduced the first parametric equalizer in early 1971. His design leveraged the high performance op-amp of his own design, the 535 series (USPTO #3727896) to achieve filtering circuits that were before impossible. Flickinger's patent (USPTO #3752928) from early in 1971 shows the circuit topology (Figure 2.) that would come to dominate audio equalization until the present day, as well as the theoretical underpinnings of the elegant circuit.

Instead of slide potentiometers working on individual bands of frequency, or rotary switches, Flickinger's circuit allowed completely arbitrary selection of frequency and cut/boost level in three overlapping bands over the entire audio spectrum. Six knobs on his early EQ's would control these sweepable filters. Up to six switches were incorporated to select shelving on the high and low bands, and bypassing for any unused band for the purest signal path. His original model boasts specifications that are seldom met today.

Other similar designs appeared soon thereafter from George Massenburg (in 1972) and Burgess McNeal from ITI corp. In May 1972 Massenburg introduced the term *Parametric Equalization* in a paper presented at the 42nd convention of the [[Audio Engineering Society. [1] (http://www.massenburg.com/gml/downloads/parametric_paper.pdf) </ref> Most channel equalization on mixing consoles made from 1971 to the present day rely upon the designs of Flickinger, Massenburg and McNeal in either semi or fully parametric topology[*citation needed*].

In the late 1990s and in the 2000s, parametric equalizers became increasingly available as Digital Signal Processing (DSP) equipment, usually in the form of plug-ins for various digital audio workstations. Standalone outboard gear versions of DSP parametric equalizers were also quickly introduced after the software versions and are typically called Digital Parametric Equalizers.

References

1. ^ ***a b c*** Strong, Jeff (2005). *PC Recording Studios for Dummies* (http://books.google.com/books?id=4-2SfnghnYwC&dq=pc+recording+studios+for+dummies&source=gbs_navlinks_s) . For Dummies. p. 25. http://books.google.com/books?id=4-2SfnghnYwC&dq=pc+recording+studios+for+dummies&source=gbs_navlinks_s.
2. ^ ***a b c d*** Louie, Gary; White, Glenn (2005). *The Audio Dictionary* (http://books.google.com/books?id=DulVm8t88QkC&dq=the+audio+dictionary&source=gbs_navlinks_s) . University of Washington Press,. p. 140. http://books.google.com/books?id=DulVm8t88QkC&dq=the+audio+dictionary&source=gbs_navlinks_s.
3. ^ Hodgson, Jay (2010), *Understanding Records*, p.73. ISBN 978-1-4411-5607-5.
4. ^ Hodgson (2010), p.73-74.
5. ^ Hodgson (2010), p.74.
6. ^ Diverse Devices. *Video and Audio Abbreviations and specialised terms* (http://www.divdev.fsnet.co.uk/abbrev.htm)
7. ^ Hodgson (2010), p.75-76.
8. ^ Hodgson (2010), p.76-77.
9. ^ Langevin EQ-251A Schematic (http://www.gyraf.dk/schematics/Langevin_EQ251A.GIF)
10. ^ Operator Adjustable Equalizers: An Overview (http://www.rane.com/note122.html)

External links

- Discriminating EQ frequencies by ear (http://www.audiocheck.net/engineertraining_bands_difficult.php)
- Calculator: bandwidth per octave N to quality factor Q and back (http://www.sengpielaudio.com/calculator-bandwidth.htm)
- EQ Condensed Overview (http://www.idc.ul.ie/idcwiki/index.php/Equalisation)
- WikiRecording's Guide to Equalization (http://www.wikirecording.org/EQ)
- Audio EQ Cookbook (http://www.musicdsp.org/files/Audio-EQ-Cookbook.txt)

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