SPL Analog Code® Plug-in
Manual

Passeq

Dual-Channel Three-Band EQ

Manual

Passeq Analog Code® Plug-in, Model Number 1040

Manual Version 2.0 – 12 /2011

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Installation

Plugin Alliance Activation

Your Analog Code plug-in must be activated in your Plugin Alliance account. You can set it up and log into your account anytime at **http://www.plugin-alliance.com**

For details about the activation process, read the Plugin Alliance Activation Manual. The PDF file is stored in the same folder of your computer like this product manual file.

Alternatively, the following web page provides the same information: **http://www.plugin-alliance.com/activation**

System Requirements and Compatibility

For details about system requirements and supported platforms or formats visit **http://www.plugin-alliance.com/compatibility**

MAC and Windows Installation

1. Check for the latest plug-in software version before installation: **http://software.spl.info/download**
2. Execute the installer file and follow the instructions.

Glossary

Host Program: program on which the plug-in is running (Pro Tools, Cubase, Logic, etc.).

M/S: Mid/Side encoded signal information as an alternative to standard left/right (L/R) encoded stereo signals. For more background information on the M/S technique read page 28 and 29.

SPL Analog Code® Plug-ins

While SPL hardware products have been fascinating audio professionals from home studio owners to mastering engineers in the world's most renowned facilities for years, the need for this technology in the form of plug-ins has also been an ever-growing demand. With the Analog Code® plug-ins we have finally accomplished our much desired goal: to transfer to the digital domain the high quality we have striven to achieve with our analog processors throughout several decades.
The first time we ever heard a software that fulfilled our expectations, one of our hardware developers said to the programmers: "you have cracked the Analog Code" — thus was coined the name of our digital products.

The most powerful passive EQ system ever

The original Passeq hardware is the first passive EQ which provides three separate frequency ranges for both boost and cut stages. One famous, if not the most famous, passive design was the Pulteq EQ from the decades of the 1950's and 60's. This EQ sported two frequency bands (low and high frequencies, or LF and HF), and had only a few switchable frequencies to offer. In contrast, the Passeq has 12 switchable frequencies per band, totaling 36 boost and 36 cut frequencies. Boost and cut frequencies are NOT identical, thus the resultant 72 frequencies per channel offer an enormous choice for the most elaborate EQ curves.
The Passeq offers for the first time passive filter control possibilities extending throughout the relevant audio frequency range—and that with an unheard of abundance of filter choices.

The Passeq Analog Code® Plug-in

The fantastic qualities of SPL's Analog Code programming faithfully reproduce the unique sound quality of the original hardware. All the complex interactions between each single filter are reproduced in every detail. One of the peculiarities of passive filter designs is that they can be seen as one big filter — the signal always runs through the whole network. Whenever the settings change, a flow of interactions takes place between the filters, providing the characteristic and unique sound of these EQs — something that will never happen with active EQ designs.

Special Features

Two Graphical User Interfaces (GUI): The plug-in list of your host program will show two entries after installation: “Passeq” and “Passeq Single.” Aside from the standard dual-channel GUI, the single GUI represents a space saving alternative showing one Passeq channel. Nevertheless, the single version can be applied to a stereo track — you simply control both channels simultaneously with one knob.

M/S Mode: as an alternative to L/R processing you can also switch to M/S mode for processing the mid and side information of a stereo panorama. Read more on page 15 (“M/S”) and 28 ff (“M/S Basics”).

Mouse wheel control for all rotary knobs

All SPL Analog Code plug-ins support mouse wheel control for rotary controls and faders. Place the mouse cursor over a rotary control and move the scroll wheel of your mouse to adjust the setting. Hold the CTRL (Windows) or COMMAND (Apple) key while moving the scroll wheel to make fine adjustments; the resolution of the mouse wheel is increased, making fine-tuning easier.

Keyboard Shortcuts

All SPL Analog Code plug-ins support format and OS specific functions for value reset, fine adjustment and mouse control. For more detailed information please refer to the host program’s documentation.

Mono, stereo or multi-channel operation

The Passeq plug-in can be used either for mono or stereo operation. You can also use the Passeq as a „Multi-Mono“ or multi-channel plug-in, as long as your host program supports this function.

Layout of Operational Elements

Initially one might be struck by the circular arrangement of the Passeq's control elements. As unusual as this first appears, the more understandable and clearer this layout becomes when one looks closer.

Along with the fact that we simply like this design from an aesthetical view, this layout makes even more sense with respect to the idea of the passive EQ concept itself: In a passive design, filters for boosting and cutting a frequency range are physically separated from each other. Reflecting this fact, the elements left of the central output control perform level cuts, while controls to the right of this central regulator serve as signal boost controls. Cut and boost switches are positioned next to the appropriate frequency band selector and frequency bands are arranged from low to high from the standpoint of both physical and frequency range layout—all in all a clear overall functional picture though without much in the way of boring routine.

Passeq Single – the space-saving single channel GUI

Allocation of Frequencies

One of the greatest Passeq design challenges was in determining the choice of frequencies, which in contrast to parametric EQ designs, are fixed or nonadjustable. One could accept standardized values from such as the so-called ISO frequencies, but such measurements stem too much either from conventional measurement standards or those from room corrections rather than choices of what may be musically more sensible. In assigning the Passeq's frequencies it was inevitable that we would rely on the nearly 30 years of experience of SPL's chief developer, audio engineer and musician, Wolfgang Neumann.

To enhance further our achieving this musical objective many audio experts and musicians were consulted regarding their favored frequencies. Among the many, David Reitzas, Michael Wagener, Bob Ludwig, Ronald Prent and Peter Schmidt offered valuable advice. From this point of departure we managed to determine that there is definite agreement among professionals about their preferred musical frequencies, and these differ clearly from the standard ISO choices.

The results also showed that the closely meshed boost and cut frequencies are important and sensible. Through them one can on the one hand focus more precisely on a certain frequency, and on the other, offer the option of influencing the Q factor (which is typically rather small in passive designs) by creating so-called S curves. An Example: Assume you wish to boost in the mids around 320 Hz, an instrument or voice level while at the same time avoiding a boost to the frequency range below it due to the small Q factor (high bandwidth) of the filter, and perhaps even lower it. In this case, let's say you choose the LMF-MHF boost band and increase the chosen (320 Hz) frequency range by about 3 dB. At the same time, you chose a 4 dB reduction in the LF-LMF cut band. The close proximity of the chosen frequencies allows you achieve an increase in the slope between the two. This is "S slope EQ-ing" at its best, and in this discipline, the Passeq is a world champion in both options and results.

LF-LMF Cut and LF Boost

The low cut frequency range extends from 30 Hz to 1.9 kHz and will be referred to in this text as LF-LMF (Low to Low-Mid frequencies). In contrast, the low boost (LF Boost) band encompasses a range of 10 Hz to 550 Hz. The maximum available increase in this LF boost band is (+)17 dB, while the maximum reduction of the LF-LMF cut band is (-)22 dB.

Optically these filter bands may be represented as having a shelving characteristic with an 6 dB slope. Passive filters do not allow for direct alteration of the slope gradient because this quality is pre-determined by component selection and not, as with active filters, by a variable value. The lowest frequencies begin here with 10 Hz, then follow with 15, 18, 26, 40 Hz, and so on. At this point one might think that such a lavish set of frequency choice in this range might be a bit overdone, as there is acoustically a rather limited amount of audio material of any real significance below 26 Hz. However, these choices are anything but arbitrary. These frequencies represent a consistent -3 dB point of a sloping down response curve. That is, the gentle 6 dB slope also allows frequencies above 10 Hz to be processed. For the hardware original, special condenser/coil/resistor filter networks have been designed for each frequency range. The choice of one or the other inductances produces differences in sonic coloration even when limited differences between frequencies such as 10 Hz or 15 Hz play a subordinate role. Along with this differing phase relationships may come into play and affect tonal color. Because modern productions often demand a definite number of choices in an engineer's options for achieving an optimal result in bass emphasis, the Passeq has been designed with a very complete set of low frequency options to insure realizing these goals.

MF-MHF Cut and LMF-MHF Boost

The midrange bands elevate the Passeq to a complete combination of filter options that classic passive designs do not offer. Both midrange bands exhibit peak filter characteristics, that is, when viewed from the boost band, the frequency curve appears as bell-shaped slopes above and below the chosen frequency range. The slope or Q-value is, again, not variable, but attuned through the choice and configuration of the passive filter‘s components for a maximum in musical efficiency, relying in the Passeq on its developer, Wolfgang Neumann‘s years of musical experience. The middle bands‘ peak structure is chosen for a clean separation of LF and HF bands. Were the choice here to be for a shelving filter design, too many neighboring frequencies would be processed, with resulting undesirable influences extending into LF and HF bands. Along with this is the simple fact that a midrange peak filter characteristic is accompanied by a more easily focused center point processing of critical voice and instrument fundamental frequencies.

The MF-MHF cut band overlaps the LF-LMF cut band by approximately an octave, with its lowest frequency extending from 1 kHz. The LF boost and LMF-MHF boost bands are set up in a similar fashion, with the lowest LMF-MHF boost band frequency set at 220 Hz and thereby 1-1/2 octaves under the highest LF boost band frequency. The maximum values of the MF-MHF cut and LMF-MHF boost band extend from -11.5 dB to +10 dB.

The overlapping band characteristics give a good idea of the available degree of precision in frequency adjustment: For example, one can boost in the LMF-MHF boost band at 220 Hz while in the LF boost band, 240 Hz can be followed by 320 Hz in the LMF-MHF boost band: The next step could be at 380 Hz in the LF boost band, followed by 460 Hz in the LMF-MHF boost band and 550 Hz in the LF boost band ...

MHF-HF Cut and HF Boost

Passeq's high frequency bands have a different layout for the cut and boost ranges: The MHF-HF cut band exhibits a (wide-band) shelving characteristic, while the HF boost band exhibits a variable Q, peak filter characteristic.
As seen above, one can also note and intensification in choice of frequencies in the high range. Here the same reasons apply as in prior cases: The individually designed and constructed coil-condenser-resistor configurations of the hardware original result in slightly differing sonic characteristics. Thus beginning at 10 kHz there are seven additional switchable frequencies. The available variable Q (ranging from Q=0.1 to Q=1.0) allows the engineer access to an enormously flexible range in high frequency boost options.

HF Boost Q with Proportional Q

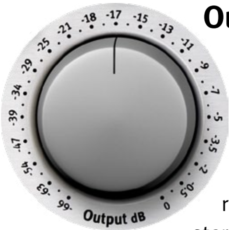
With the proportional or variable Q principle, boost control settings would apply only if the HF boost Q were to be set at Q=1.0 (control set fully clockwise). Were the value to be reduced (thus increasing the bandwidth), the boost would also be reduced. This can lead to a situation wherein, for example, a HF boost Q setting of 0.1 and a boost of 3 dB would result in effectively no audible boost in the chosen frequency—at this value the Q value resides at about 0.3 dB. With this Q value, don't hesitate to turn up the HF band boost control to its full 12.5 dB setting—this results in an actual overall increase of around 3.5 dB. Narrower Q settings, for example, to 0.6, result in further level boosts again.

The advantage of proportional Q as compared to constant Q designs rests with the musically superior way it functions. The wave energy which resides below the bell curve remains essentially the same and in the process, retains the balance of high frequencies in relation to the entire frequency spectrum as one experiments with varying Q values. While it is true that one must think independently of the scaled HF boost dB values in such cases (because these only apply to a value of 1), the result is a simpler, more musically sensible and worthwhile way to work that does not require continual additional corrections.

MHF-HF Cut

The MHF-HF cut band is similar to a shelving filter that can reduce higher frequencies in a wide bandwidth. It is appropriately wide, beginning with 580 Hz and extending to 19.5 kHz, a range of over 5 octaves and overlapping the lowest, LF-LMF cut band by just about two octaves. With it one can lower a very wide bandwidth and with the peak mid range filters further reduce—or raise—specific ranges. The process can result in the creation of very interesting curves. Here the maximum cut is -14.5 dB, while the maximum boost reaches +12.5 dB.

The Passeq is not limited to any one particular kind of application, and, for example, is also especially well suited to processing individual instruments in recording sessions. In such cases the wide downward reaching MHF-HF cut band may be play an exceptional role. Individual instruments can easily be cut upwards, either to give them a more compact sound or when higher frequencies might be supplied from different microphone—or because the mix simply suggests it.



Output Control

The Output control serves as output level regulator. If the output level increases due to frequency boosting, it can always be reduced again to match the input level. When you start using the plug-in, the control is set fully clockwise, i.e. at 0 dB. From there you can reduce the level to a maximum of -66 dB. The scale starts with a very fine resolution to allow for smooth adjustments, especially in the first half of the control range.

Settings

The four SETTINGS buttons allow you to save all your settings with a simple mouse click. As soon as you click on another SETTINGS button, the current settings are saved under the previously active preset. For example: In the image shown here, all parameters would be saved under preset “A” if you were to click on another button.
Any previously saved preset can be recalled with a simple mouse click on the corresponding button; you can then use or edit the settings. If the host program allows it, the presets can also be automated so you can use different settings at different points. As long as you work in a specific session of the host program and the plug-in is installed, the settings are saved and can be recalled afterwards. When opened, the plug-in loads the active preset settings instead of the default settings. If you remove the plug-in from the host program all presets are lost. To erase all presets at once you can remove the plug-in from the host program and then reinstall it.

Channel Switch

Two illuminated switches in the center of the standard GUI front activate or bypass processing for the left or right channel. On the Passeq Single GUI, the right switch is always grayed out; the left one always activates or deactivates the Passeq Single, regardless of whether processing is mono or stereo.

M/S and Link

The M/S and LINK switches are only available in the standard GUI, because separate operation of each channel is required to use them. **IMPORTANT:** Usually, the LINK switch should be deactivated when you activate M/S — otherwise the settings for the mid signal are applied to the side signal as well.

M/S

As an alternative to L/R encoded stereo signals, there is one technique that is particularly useful for signal processing during production: M/S. "M" stands for Middle (or Mid) and "S" for Side, which means that signals are separated from the middle to the sides, instead of from left to right.
The M/S switch activates an M/S encoding of the L/R signals. Now you can process the mid information with the left channel and the side information with the right channel. M/S encoding is only used for processing; decoding into L/R format is done before the signal is output. Note that both encoding and decoding is lossless.
The use of a mid (M) and a side (S) signal instead of the usual L/R signal results in a much wiser musical processing. High-energy Mid signals (vocals, snare, bass guitars, etc.) can be easily separated from Side signals (guitars, keyboards, cymbals, etc.). When processing sum signals, M/S encoding is often the best option to be able to target single elements within a mix. Also refer to "M/S Basics" on page 28 ff.

Link

The LINK switch couples both channels. This ensures the same settings for the two channels, and it takes half the effort to set them up.
The left channel always controls the right channel. If you activate the LINK mode, the current settings on the right channel are not overwritten, even if they are different. Settings are only transferred once LINK mode has been activated.

Using Equalizers

In the arenas of recording and mixing one can generally distinguish between two main goals in applying EQ: The first is sound correction, or sound design through processing of individual channels while the second may be improving their separation or presence in the mix. In the overall recording process there may be deficiencies due to technical problems, for example, noise or bleeding of neighboring instrument sounds that detract from the natural quality of the desired instrument. Through frequency response characteristics of a microphone or phase shifts due to reflections, energy at certain frequencies can be reduced or get lost, denigrating the original sound quality of an instrument. EQ is probably the most important tool to combat these problem areas. Moreover, an instrument‘s sound can also be accentuated or emphasized—to the point that this becomes in its own right a creative sonic activity with a production made only possible by the employment of EQs and their special characteristics.

Basic Approach

While we would never assume that in creative and artistic work there should be absolute rules, and this also applies to work with EQ: There is no such thing as "The Voice" or "The Kick Drum" or "The Piano". The following is thus offered strictly as a basic orientation or starting point for such work, and should not be misconstrued as dogma or any other kind of absolute. Nonetheless, in order to achieve sometimes hard-to-define goals when applying EQ, it really is important to be aware of and be able to use a few accepted basic musical and technical guidelines.

EQ Yin & Yang

This section on "EQ Yin & Yang" reproduces thoughts and verbalizations by Bob Katz, whose superb Focal Press book based on a series of lectures entitled, "Mastering audio, the art and the science", we highly recommend.

In Chinese philosophy, Yin and Yang describe unconditionally bound opposites within some kind of unity, which in turn, both complement and conflict with one other. This idea also provides an insightful analogy to the understanding of the connection between music, harmony, fundamentals and harmonics. This mutual bond and interaction between such opposites creates inevitable and mutual reactions and repercussions in the other whenever something occurs to one. Here are a few examples:

- A reduction in the lower middle range around 250 Hz can have a similar effect as an increase in the presence region of 5 kHz.
- Added energy in the very high region (15-20 kHz) can create the impression of having made the bass and lower mids thinner.
- Adding warmth to a voice will reduce its mix presence.
- Working with EQ and this Yin and Yang principle means ideally to consider always such implied repercussions of work in one frequency—for example, that in working to enhance warmth, that one might want to avoid losing presence.
- Harshness in the upper middle to lower high range can be countered with more than one approach: A harsh trumpet section may be improved through a reduction around 6-8 kHz, or with an increase at around 250 Hz. Both of these measures result in a warmer sound, but the decision of which to use should depend on which of the two also works best in the entire mix.
- Moreover, one should never forget how easy it is, while working intensely with isolated elements of a mix, to fall into the trap of forgetting how such elements can influence, for better or worse, the rest of the mix.

First control levels, then apply EQ

Badly adjusted levels often induce us to misuse EQ in misguided efforts to correct them. As soon as one has the feeling that he or she needs more that 6 dB in EQ (boost), one should investigate thoroughly whether or not initial levels have been set properly.

Using Equalizers

First cut, then boost

“The ear” is more used to energy reductions in a frequency range, thus boosts attract more attention. That is, a 6 dB boost is perceived to be similar in amount to a 9 dB cut. Therefore when wishing to emphasize one frequency, it is typically better first to consider a reduction in others. The result will bring more transparency and clarity as well as reduce possible unwanted coloration of the signal.

Reducing bleed from other instruments or noise outside an instrument‘s range

Wide band filters setups should be chosen with threshold frequencies in ranges from one-to-two octaves above or below the highest or deepest instrument‘s frequency. Example: To eliminate cymbal bleeding in a kick drum recording, one should try a setting from about 10 kHz with a 10-15 dB cut.

Reducing bleed within an instrument‘s range

The main frequencies of the bleeding instrument should be reduced as far as possible while avoiding to alter the natural sound of the main instrument in an unnatural way.

Boosting harmonic frequency levels

Harmonic enhancement is one of the foremost techniques for increasing the clarity and definition of an instrument. An overview for three typical instruments:
Bass – 400 Hz: Bass lines will be accented
Bass – 1500 Hz: More clarity and attack sounds
Guitar – 3 kHz: Clearer attacks
Guitar – 5 kHz: Brighter, more brilliance
Vocals – 5 kHz: More presence
Vocals – 10 kHz: Brighten up
Note that each instrument will have at least two frequencies where EQ can achieve a greater clarity or brilliance.

Boosting fundamental levels

Inexperienced audio engineers will often first try to make corrections by boosting fundamentals, something which in fact should be the last thing one considers. Boosting fundamentals typically lowers clarity and produces a muddy sound. If two instruments are playing the same part and thereby produce the same fundamental, raising these levels will lead to a decrease in the sonic difference between them, (i.e., will make the two instruments sound more alike and lower their intelligibility in the mix). This is also true when two instruments play similar parts in the same key.
Exception: When an instrument sounds thin or small, boosting the fundamental can help. Or perhaps a microphone was poorly placed or the harmonics had been raised excessively through EQ. Finally, increasing fundamental levels can also play a constructive role when instruments play alone or as soloists with others in the background.

Cutting fundamental levels

Cutting fundamental frequencies provides for a perceived increase in harmonics and is therefore an effective alternative to boosting harmonic levels. This is a common practice in Rock/Pop productions that can be effective in all musical recording genre. Three examples:
Bass, Reduction at 40 Hz: limits boominess, increases presence.
Guitar, Reduction at 100 Hz: limits boominess, increases clarity.
Voice, Reduction at 200 Hz: limits muddiness.

Emphasis of an instrument‘s main frequencies

For this purpose a bandwidth of 1 and 1/3 octaves is generally a very good starting point—in other words, this range best encompasses that of most instruments‘ frequency spectrum. This can be somewhat narrower with percussion instruments, while it is recommendable to consider a wider bandwidth for melody instruments such as voice or bowed strings. The boost value should remain between 3 and 6 dB.

In the mix—or not?

The more an instrument is placed "outside" a mix (resp. above or in front of a mix), the more natural its sound should remain. When already embedded in a mix, main frequencies should on the other hand be processed with a higher dB value but lower bandwidth. An example: A boost of 3dB at 5 kHz may serve to make a voice track clearer and much more present in front of a mix, while when embedded in the mix, a 6 dB boost with less bandwidth may be more useful.

Splitting frequency bands to reduce masking effects

In order to separate two instruments whose sound lies in the same range, one may choose to process frequencies that are a half an octave from each other. With a bandwidth of a half octave and 3 dB boost, one can achieve clarity and instrument differentiation. The higher frequency should by applied to the instrument which sounds brighter or more brilliant.

Complementary filtering

One of the most difficult problems in mixing instruments is the masking effect. Loud instruments cover others when their frequencies lie in the same range. It can be very frustrating to discover that a terrific sounding instrument track suddenly sounds boring when added to a mix.

Of great help here can be an application of the above-described frequency range separation and processing through complementary signal filtering. In the process specific frequencies of one instrument should be reduced with narrow bandwidths while increasing the same frequencies of other instruments. This involves boot and cut values between ca. 3-6 dB.

Classic conflicts of this type happen, for example, between kick drum and bass or between lead and background vocals, and these are perfect circumstances for applying complementary filtering to avoid masking problems:

- Kick Drum/Bass: A reduction of the kick drum between 350 and 400 Hz and an increase in the same bass frequencies will reduce the cardboard sound of the kick drum while lending the bass more presence.
- Lead/Background Vocals: A cut between 3 and 4 kHz in the background voices gives them a needed airy quality, while boosting the same lead vocal range allows it to come through with more clarity.

Processing Examples

Here we provide approximate values which may expand to adjacent areas.

50 Hz – cut: Reduces boominess in all lower instruments (basses, kick drums, toms). The implicit increase of the relative level of harmonics improves the presence of bass lines.

50 Hz – boost: Lower frequency instruments sound fuller.

100 Hz - cut: Limits boominess, greatly increased guitar clarity and limits sustain with Toms.

100 Hz – boost: Firmer bass sound for all low frequency instruments, adds more warmth to piano and horns.

200 Hz – cut: Less muddiness with voices and middle instruments, while helping to eliminate the “gong” resonance with cymbals.

200 Hz – boost: Fuller sound for voices, snare drums and guitars.

400 Hz – cut: Limits hollower sound qualities in lower drums.

400 Hz – boost: Clearer bass lines.

800 Hz – cut: Diminishes the “cheap” sound of some guitars.

800 Hz – boost: Noticeably clearer, punchier bass lines.

1.5 kHz – cut: Reduces an uninteresting sound in guitar tracks.

1.5 kHz – boost: Clearer, cleaner basses.

3 kHz – cut: Hides badly tuned guitars or poor intonation.

3 kHz – boost: Better bass guitar attacks, more attack with electric and acoustic guitars, snares and other percussion as well as lower piano parts, more voice clarity.

5 kHz – cut: Softens thinner or tiny sounding guitars.

5 kHz – boost: Improves voice presence and brightens guitars, gives more attack to low frequency drums, piano, and A-guitars.

7 kHz – cut: Reduces sibilants.

7 kHz – boost: Provides more attack with percussive instruments.

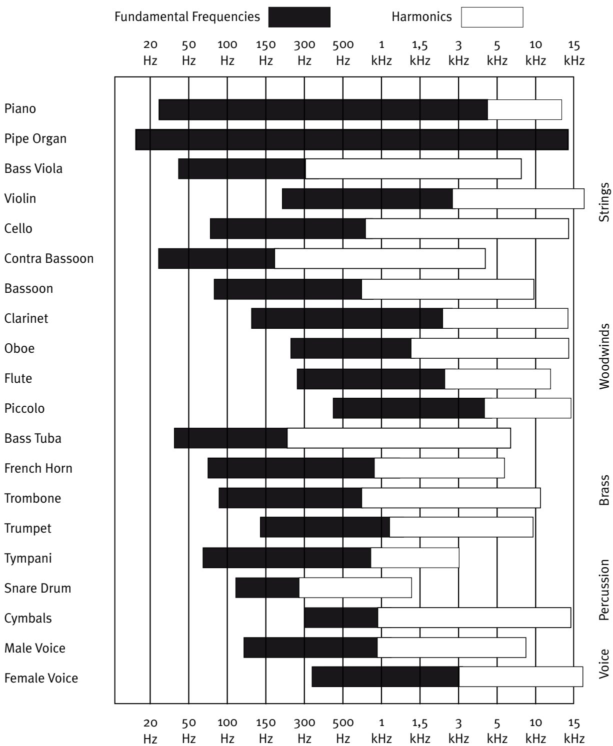
10 kHz – cut: Also reduces sibilants.

10 kHz – boost: Brightens voices, similarly brightens guitar, piano and harder cymbals.

15 kHz: Boosts in this range brighten most sounds, but be careful with hidden dangers such as emphasizing noise, hiss and/or creating excessive sibilance. The rule always applies: Before reaching for the knob to boost, first try cutting elsewhere for accentuations.

Classical instruments and their frequencies

A symphony orchestra presents a kind of ideal paradigm of a balanced, wide-spectrum instrumental sound canvas. It is therefore only sensible to consider its sound as an orientation point also for other musical genres—it will definitely not harm a Rock or Pop production to employ such an orientation to achieve a comparable balance and proper distribution of mix elements in the latter.



Frequency and Energy

In general, a frequency prescribes a number of events in a time interval. The per-second cycle of a wave form is given in Hertz (Hz). Lower tones produce longer waves and higher, waves of shorter length, and the higher the frequency, the higher the tone. The higher the amplitude of a wave, the higher its energy level and in turn, the louder it is perceived.

Tone and Sound

In the area of music, a sound event is referred to as a tone. Such a tone is complex: it is comprised of different frequencies, each at a different energy level.

In analyzing the components of a naturally produced tone (such as those created by a real instrument or voice), we see the following ingredients: A natural tone is comprised of a lowest pitch or fundamental along with many additional higher components called harmonics. The arrangement of these pitches is called the harmonic series, which includes the entire group of frequencies from fundamental to higher harmonics and is called the frequency spectrum of a tone.

As the lowest pitch, a fundamental determines the basic frequency and its perceived pitch. The frequencies of the harmonics are multiples of the fundamental frequency and determine the specific sound of a tone (that is, whether it sounds like an instrument, voice, etc.).

Should one wish by electronic means to record, process and play back a given tone, it is crucial to maintain the accuracy of the frequency spectrum if one wishes to be able to recognize it later as the original. Just as important is the aspect of maintaining the original energy levels of all frequencies it is composed of.

In producing a tone, the distribution of energy within the frequency spectrum is further and decisively influenced by the acoustic environment through the mixing of direct and reflected sound. The energy relationship between fundamentals and harmonics is different between direct sound and that which is reflected (for example, harmonics of a reflected frequency spectrum may have measurable more energy), and this can change a tone‘s perceived sound. Later, when a musician has the impression that a recording is not true to the original, that he or she has either played or sung, an important consideration to make is to examine the resultant frequency spectrum.

Sound Correction and Sound Design

Along with acoustic influences of a recording ambience, it should also be understood and accepted that, to say the least, there are definite technical limits to recording and playback that may strongly influence an end result. In the first decades of electronic recording, the principle influence on the quality of such recordings centered on the choice and placement of microphones. The first Equalzers were used to combat technical and acoustical problems such as insufficient frequency response from microphones and loudspeakers or even inadequate relationships in room acoustics that needed to be corrected or brought into balance. The goal was always to bring into balance as much as possible—and maintain—the correct frequency spectrum of an originating tone source.
The introduction of multi-track recording in the 1960’s brought a fundamental change in the way recording was done. Instruments intended for one production could be recorded in separate sessions and at different times. The mix from many individual recordings—at first in only four tracks—nonetheless added a new problem of offering ways to add further sonic quality through further processing of individual tracks, because with each copy a track’s quality was reduced.

This introduced an entirely new function for EQ, for example, the emphasizing of particular instruments in their tracks to prevent them from being lost in a mix by altering certain parts of their frequency ranges. Along with this ability to emphasize specific tonal qualities appeared the added capacity to indulge in even more creative work through much stronger or even exaggerrated processing of a sound and thereby lend it even more presence in mixes. Without a doubt, the increasing popularity of electronic tone production has played a large part in the further development of EQ filtering as a creative element in audio production.

Frequency Filters

As a rule almost everyone of us has first made an aquaintance with frequency filtering through our listening to home stereos. Such elementary kinds of filters are simple amplitude-based filters: When one turns a bass control clockwise, one hears a general or overall increase in bass frequency energy. But with the explanation above on the composition of a complex, natural tone, it is clear that such a low frequency control does not only influence the energy of the fundamental frequency, but also always the sound of a tone—the relationship between energy of the fundamental and harmonics frequencies is changed. Typically amplitude-based frequency filtering boosts or cuts the energy of a specific audio frequency band. In such processes it is possible to employ filters with design and function that are very different from each other: Depending upon the technical construction, such filters may, for example, process only high or low frequencies in certain way.

Filter Types

There are two types of filters used in the Passeq: wide-band filters which are comparable to shelf-filter characteristics and bell-formed peak-filters with narrower bandwidths.

Shelf Filters

A shelf filter increases or decreases the energy of all frequencies above or below a chosen frequency. Depending upon the direction of processing one refers to high frequency (HF) or low frequency (LF) shelf filters. Beginning with the threshold frequency, the frequency band is boosted or cut much like a shelf. The maximum boost or cut achieved at the point furthest from the threshold frequency. The threshold frequency is usually about 3 dB less (with the overall increase set to maximum). This gives the typical rising form of the shelf filter's response curve.

Peak Filters

A peak filter boosts or cuts a chosen frequency‘s energy with a maximum amplitude and a definable frequency range around this frequency with a fall off of up to 3 dB to both sides. The chosen frequency with the maximum amplitude is called center frequency—it takes place in the middle at the peak of the response curve. The response curve forms a bell, thus peak filters are also often referred to as bell filters.

Bandwidth

The width of a frequency range or band is musically defined in octaves. The technical counterpart to this is the “Quality” of a filter, and the abbreviated “Q” is the most common value for the bandwidth of a filter. A high Q value means a narrow bandwidth while a smaller Q factor corresponds to a wider one:
Bandwidth 2 Octaven: 0.7 Q
Bandwidth 1 1/3 Octaven: 1 Q
Bandwidth 1 Octave: 1.4 Q
Bandwidth 1/2 Octave: 2.8 Q

M/S Stereophony

Our ability to identify the direction and distance of a sound source is the essence of spatial hearing. The human ear can identify level and time differences from ear to ear very precisely and use that information to localize sound. At frequencies up to 1500 Hz the ear analyzes basically time differences to localize sound, while above this frequency it uses level differences.

Our hearing provides excellent conditions to apply room information even to artificially generated sounds. Regardless of the deficiencies and differences that loudspeakers and headphones might present during playback, the human ear needs only the signals to be codified in, at least, two channels, in order to be able to identify time and level variations, which result in spatial hearing.

This sort of recording and playback that includes spatial information is known as stereophony (from Greek stereos = solid or three-dimensional). The resulting stereo image is called panorama. Besides the two-channel stereophony there are several other formats of stereophony. The common conception of „stereo" as a two-channel recording is thus incorrect.

Equally incorrect is the concept that the encoding of a stereo signal is always done in a right and a left channel. This idea is based on the fact that we have a right and a left ear and that all two-channel recording and playback systems use the same right/left format. It is also not true that all recordings are made with a microphone for the left channel and a microphone for the right channel.

The differences between the most important microphone techniques have much more to do with level and time differences. Due to the advantages and disadvantages that each technique provides, more often than not they are combined during production to achieve L/R playback.

While there are several stereo techniques that can be applied during miking, for signal processing during production there is only one technique that is actually useful: M/S. „M“ stands for Middle (or Mid) and „S“ for Side, which means that signals are separated from the middle to the sides, instead of from left to right.

M/S can be actually applied during recording: two microphones with different polar patterns record direct and spatial information. Besides the microphone technique, M/S can also be used as an alternative stereo encoding for signal processing, which means that signals do not necessarily need to be recorded with the M/S microphone technique to be able to apply M/S encoding afterwards. In fact, M/S encoding can be generated from L/R encoding by summing and subtracting signals:

$M = L + R, S = L - R$

The sum of the left and right signals in the Mid signal corresponds to the mono signal of the L/R encoding. The Side signal is also created from the L/R signal by inverting the polarity of the right channel. The sum of phase-inverted signals results in the cancellation of mono information in the signals summed; thus, the Side signal is made up of the differences between L and R. The detailed formula may be clearer (the minus sign stands for the phase inversion):

$M = L + R, S = L + (-R)$

It is also possible to create a L/R signal encoding from an M/S encoding by summing and subtracting the signals, what is usually called M/S decoding:

$L = M + S, R = M - S$

Mathematically, the sum and subtraction of signals guarantees a lossless conversion from L/R to M/S and back to L/R, which is a very important aspect for using M/S encoding for signal processing.

Passeq
SPL Analog Code® Plug-in

Manual

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