Q10
Setup Guide

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Introduction

Because of Q10's 10-section filter topography and its facility for simultaneous control of several EQ bands or parameters, it may be used to create special effects and to produce spectral manipulations that would either be extremely difficult or impossible using conventional equalizers. Q10's 'select and drag' control interface makes the control of even very specialized EQ settings very straightforward, but specialized EQ settings can be very time consuming to create. A complete Q10 stereo filter setup involves the fine-tuning of 60 filter parameters, and in a busy commercial environment, there is always the temptation to manage without a certain effect, simply because there's no time to spend setting it up. That's where the Waves Q10 Setups Library can help you.

This Library contains a great many filter setups, optimized for specific applications, and these may simply be loaded in and used just as they are, or fine tuned to the job at hand. Many of the settings make use of Q10's multiple selection facility allowing a complete filter setup to be adjusted in one click or drag.

This manual describes the various filter setups in the Library and provides information on how they might be used, and on how they may be adjusted or modified. This manual has been termed a 'Tutorial Manual' because it does more than furnish minimal instructions; it helps users unlock the power and potential of the Q10 to achieve the ultimate in creative and corrective EQ effects.

Using these setups, the Q10 is shown to be more than just a conventional parametric equalizer, it becomes a unique effects unit and processor.

All effects (with a few noted exceptions designed for use with sample rate conversion) will work at both 44.1 and 48 kHz sampling rates.

Please let us know how this library serves you, and of other setups you'd like to have included in future releases.

Operational aspects

Clipping distortion & Overload

The input and output gains of setups are generally adjusted so that the overall energy gain is unity to permit direct A/B comparisons between sounds in effected and bypass modes in preview. However, be warned that with audio files running to near peak levels, this can sometimes result in clipping or overloading at the output, causing audible distortion. Any overloads will be registered by the overload indicators above the level meters. Reducing the input levels by around 3 dB will reduce the risk of overloading.

If overload has occurred on a processed audio file, you should undo the processing, reduce the input gain, and process again.

The input gains in most setups are generally adjusted so that for signals with a normal frequency distribution, the internal processing of the EQ will be free of internal clipping distortion. Most of the time, overload distortion will occur only at the output and can be avoided by pulling down output gain. However, to be absolutely safe when processing signals with an abnormal frequency content, it is better to perform any needed gain reduction using the input faders.

An exception to this rule is the intentional distortion effects!

Sampling rates

Except where specifically stated otherwise, the effects here were devised primarily for use with audio files sampled at 44.1 or 48 kHz. Many of the setups will not work in the intended fashion at other sampling rates, especially at significantly lower sampling rates such as the 11 or 22 kHz multimedia sampling rates.

(In the older Q10 version 1.0, the setups will not load correctly at sampling rates below 44.1 kHz, and the graphical display will be incorrect.)

The setups have been optimized for use with files sampled at 44.1kHz. If files of a lower sample rate are processed, the filter frequencies will be scaled down proportionally to the sampling frequency. For example, all filter frequencies will be halved when the setup is loaded to process a 22.05 kHz sampling rate file; the new frequencies will be displayed correctly in the frequency value windows and on the graphical display.

Effects that will work at all sampling rates include EQtools, Supernotch, Superparametric EQ, Harmonic Combs, and Pseudostereo and Crossover effects, although their actual frequencies will be changed. All but some pseudostereo effects may be altered in frequency to compensate for the difference in sample rate as described below.

The bandlimiting and distortion effects will generally come out at the wrong frequencies at lower sampling rates, but some (including **Q10 telephone**) may be re-adjusted to restore the correct frequency range.

Important control information

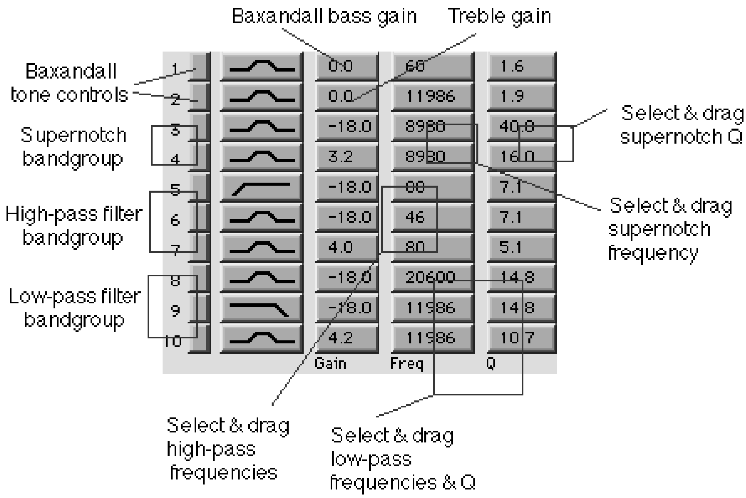
Select-and-drag control

The Q10 permits a large number of equalizer parameters to be 'linked', i.e. controlled at the same time with a single movement of the mouse. See the main Q10 manual for more information on this facility.

For most adjustments on stereo material or on mono inputs, the stereo channels should be linked.

Example

Load the Q10 EQtools, and click the 'Open' button. The screen shown below should appear.



The select and drag options and controls of the setup Q10 EQtools.

Note that many of the setups use several Q10 bands to achieve a single filtering effect.

For example, the **Q10 EQtools** setup file when loaded provides four different equalization tools. Bands 1 and 2 provide the respective bass and treble of a Baxandall equalizer while Bands 3 and 4 provide a 'Supernotch' filter for removing unwanted narrow frequency bands. Bands 5, 6 and 7 provide an adjustable steep-cut, high-pass filter for removing lowbass frequencies. Bands 8, 9 and 10 provide an adjustable steep-cut low-pass filter for removing high treble frequencies. Each bandgroup may be adjusted independently of the others.

For example, with the 'Supernotch' (Q10 bands 3 & 4) in, its frequency may be altered by selecting the frequency value windows of both bands 3 & 4, clicking the mouse within one of these, and 'dragging' the frequency value to either side. The width of the Supernotch can similarly be varied by selecting the Q value windows of Q10 bands 3 & 4, clicking the mouse within one of these, then 'dragging' the Q value to either side.

The frequency of the high-pass filter (Q10 bands 5, 6 & 7) may similarly be altered by selecting the frequency value windows of these three bands, and then click-and-dragging the frequencies to either side from within one of the selected value windows.

In all select, click and drag operations, the operation is applied to both stereo channels provided that Strap mode is selected. If Strap mode is not selected, then the operation is only applied to the channel (left or right) selected at that time.

In the **Q10 EQtools** setup, the frequency of the low-pass filter (Q10 bands 8 to 10) may be altered by selecting the 3 frequency value windows *and the three Q value windows* of these 3 bands 8 to 10, and then click-and-dragging (to either side) the frequencies and Q's simultaneously from within one of the selected value windows.

Note that for the low-pass filter, dragging the frequency values alone will not give a flat response before cut-off as the frequency is varied.

Editing of setups by selection and pasting

Any selection of settings of buttons and value windows can be copied over to any other by selecting, pressing the 'c' key on the keyboard, select the destination and pressing the 'v' key on the keyboard.

For example, in the **Q10 EQtools**, you may wish to use two Supernotches at different frequencies, but to include no Baxandall controls. Select both bands 3 & 4, press 'c' on the keyboard, select both bands 1 and 2, press 'v' on the keyboard, and you've pasted a second Supernotch to bandgroup 1 & 2.

Select and paste lets you create your own setups for fine-tuning. For example, you may paste six bands of pseudostereo from six adjacent bands of one of the 10-band pseudostereo setups, such as **Q10 PS10band**, and combine it with two Baxandall EQ bands copied from **Q10 EQtools**. Then you might add a Superparametric presence peak copied from two Q10 bands of **Q10 PresenceA**. The resulting setup can then be saved for future reuse and modification.

Chapter 1 - Pseudostereo

About Pseudostereo

Pseudostereo effects are intended to simulate a spacious or spread stereo sound from mono source material, and can be used to enhance old mono material, as well as to spread and 'liven up' mono components such as vocals, mono ambiences or mono instrumental lines within a stereo multitrack mix.

All the pseudostereo effects work by panning different frequency components of the sound alternately to either side of the stereo stage. Pseudostereo effects differ in the precise way stereo position varies with frequency. Most of the setups are characterized by the number of pseudostereo frequency 'bands' employed - the number of alternations of stereo position between left and right as frequency is varied across the audio band.

All the pseudostereo setups are intended for use with mono source material and are designed to minimize tonal changes in the original mono sound by retaining a flat total energy response from the two stereo loudspeakers.

The sound of all pseudostereo effects may be auditioned in stereo in Preview or Playback modes (in host applications where these modes are relevant) either with mono files or with stereo files having identical left and right channels ('double mono' stereo files). However, be aware that because of the processing limitations of Sound Designer II, if you wish to process a mono file to a pseudostereo stereo file, you must start off with a double-mono stereo file, otherwise, the processed file will still be in mono. You may convert a mono file into a double-mono stereo file rapidly in Sound Designer 2 by selecting 'Save a Copy...' in the file menu, then selecting Sound Designer II, 16-bit, stereo mode, entering a new file name, and then clicking the 'Save' button.

If used on stereo mixes, then the pseudostereo effects will work properly only for sounds in the center of the mix, with a risk of unpleasant tonal alteration to sounds panned to either side of the stereo stage. However, the effect may sometimes work well applied to stereo mixes where all the important sound sources are near the center, and where the stereo components are mainly reverb ambiences or spread sounds such as crowd noise, stereo rain sounds, or ADT'd sounds where the direct and delayed sound is applied at similar levels to opposite channels.

How well this will work for a specific piece of program material can only be determined by listening. When doing so, check for possible losses of frequency components in sounds at the two sides. The chances of success when applied to stereo mixes is increased if the 'narrow' versions of pseudostereo setups are used. However, these will provide a less dramatic spread of central mono sounds in the mix.

The principles of pseudo stereo are simple. We've already stated that different audio frequencies are panned alternately to the left and the right, but while the principle is simple enough, adjustments need to be very precise; even a very small inappropriate adjustment can ruin the general tonal balance. Unless you have plenty of time, and the necessary experience, to do fine tuning, it is not recommended that the tonal controls of the pseudostereo effects be adjusted individually, but that adjustments be confined to the select-and-drag options recommended in the following. These provide a wide range of adjustment in the pseudostereo effect while retaining the best available tonal neutrality.

It is important that all adjustments of pseudostereo setups should be made with the stereo channels strapped. If the Strap button is not lit up, click on it.

There are many basic varieties of pseudostereo effect available, each of which is also provided in a stereo reversed version.

Subjective aspects of pseudostereo

Pseudostereo effects have historically gained a very bad reputation, due to the very poor sound of most early attempts at this effect. Though the pseudostereo effects provided as setups, and their modifications as described below, are designed for maximum tonal neutrality, some subjective tonal effects are inevitable whenever the stereo image of a signal is manipulated. However, Q10 is able to handle pseudostereo processing with far fewer side effects than traditional systems.

Phasiness. Because the Q10 is a parametric equalizer with conventional so-called 'minimum phase' behavior, any non-flat setting causes some phase shifts as well as the frequency dependent amplitude gain variations seen on the display graph. When creating pseudostereo effects, this inevitably means that there will be phase differences between the stereo channels, which can sometimes be audible as an unpleasant 'phasiness'. The narrow setups will generally be found to be less phasey, and phasiness effects may also be found to be less noticeable in the 2-band setups.

The wide range of setups enables the user to select whichever version works best with the source material being used.

Mono Compatibility. Most of the pseudostereo setups provided are broadly mono-compatible, without suckouts or wide excursions in frequency response, although some setups will not give a completely flat frequency response when heard later in mono. Generally speaking, the historic, 10-, 14- and 18-band setups are fully mono compatible, and the tunable, 7-band and 10-bandX setups are generally the least mono compatible, but still usually acceptable. If mono compatibility is important, the 'narrow' versions of the setups will generally be more mono compatible than the normal. In all cases, a very wide pseudostereo sound can be obtained by clicking the polarity reverse button on the right channel only, but be aware that this increases phasiness and may lead to poor mono compatibility and low mono level. Again, this may be acceptable for background 'ambiences' accompanying a main dialog or vocal.

Headphone Results. It is very important to be aware that the subjective effect of stereo is very different over headphones and over loudspeakers, and it should not be assumed that a desired pseudostereo result over loudspeakers will automatically give a similar desired effect over headphones. If headphone listening is an important market for a mix, (e.g. on Walkmans), the results should be checked on both loudspeakers and headphones. This being said, the headphone results are usually found to be very acceptable, adding greatly to a sense of spaciousness.

Stereo Monitoring. While on the subject of monitoring, it is wise to check that the monitoring loudspeakers used are set up so as to give a good stereo imaging effect if subjective judgements on the pseudostereo are to be reliable. It is particularly important to listen in a precisely central symmetrical listening position to assess pseudostereo effects properly.

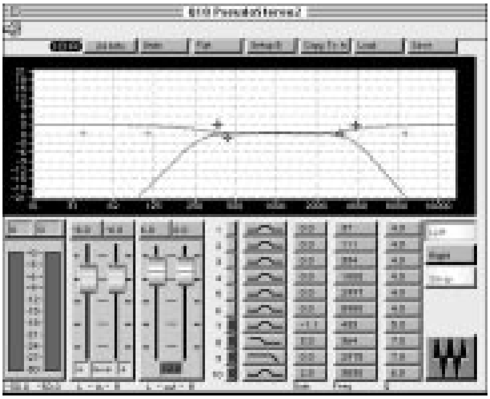
Gain settings and Overload. As provided, the input and output gain settings of the pseudostereo setups are such that the overall energy gain from input to output is unity, which permits a direct comparison between the bypass mode and the pseudostereo mode without gain changes. However, because the pseudostereo effects pan some frequency components mainly to one channel rather than splitting equally between two, this can sometimes result in channel overloads on the output in that channel. Usually, overloads can be avoided by pulling down both output gain faders - a further gain reduction of 3 dB will almost always avoid any overload problems, although the amount needed must be determined on a case-by-case basis.

Narrow and Reverse Setups. The reason for providing narrow, reverse and reverse narrow versions of setups is that these variations are often useful. For example, if a dominant frequency component is panned to the wrong side of the stereo stage, using the reverse version of the setup will pull it over to the other side. Also, if two mono sounds are both pseudostereo processed with the same setup, the result will simply be the same as the pseudostereo effect applied to a mono mix of the two, whereas if one is processed with a reversed pseudostereo setup to the other, the 'diversity' of a true stereo effect will be largely retained.

It is a good idea when processing several sounds in a mix with pseudostereo effects not to use identical pseudostereo processing on all, but to use reverse setups and different numbers of bands to create more variety of effect. Narrow setups are particularly recommended for material where an exaggerated spread should be avoided, notably speech. Narrow setups also tend to give less 'phasiness' and better mono compatibility.

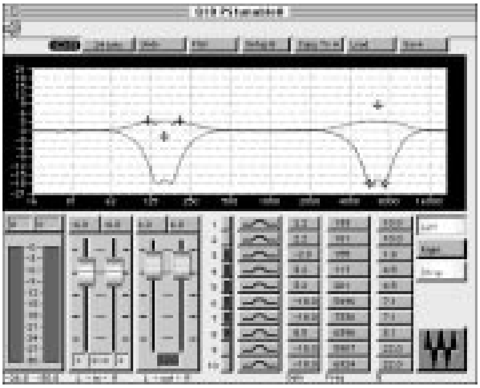
Basic Varieties of Pseudostereo

The basic varieties are:



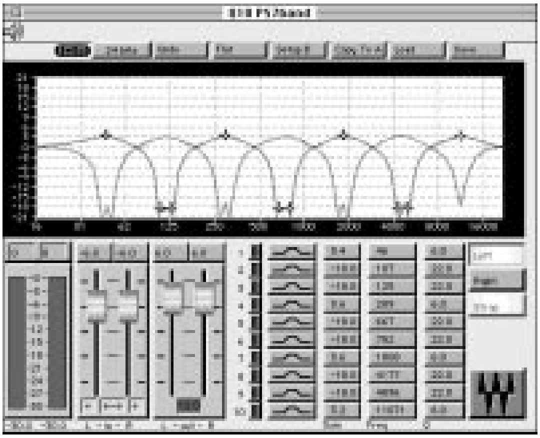
Q10 PseudoStereo2
Q10 PS2reverse (not pictured)
Q10 PS2band2 (not pictured)
Q10 PS2reverse2 (not pictured)

2-band pseudostereo effects, panning bass and treble to opposite sides of the stereo stage. A 6-band conventional graphic equalizer is also provided.



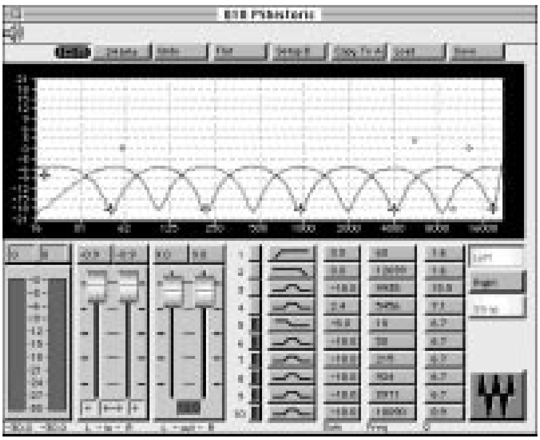
Q10 PStuneableA
Q10 PStuneableAreverse (not pictured)
Q10 PStuneableB (not pictured)
Q10 PStuneableBreverse (not pictured)
Q10 PStuneableC (not pictured)
Q10 PStuneableCreverse (not pictured)

These six are very highly tunable pseudostereo setups where individual bands can be panned to the left or the right to customize the stereo panning of different sound components.



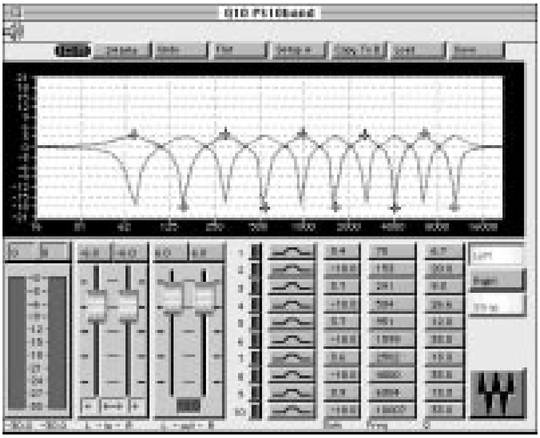
Q10 PS7band
Q10 PS7reverse (not pictured)
Q10 PS7band2 (not pictured)
Q10 PS7reverse2 (not pictured)

A 7-band pseudostereo effect, panning each of seven bands alternately to the left and to the right of the stereo stage. This effect is very wide, but not entirely mono-compatible. A highly adjustable effect.



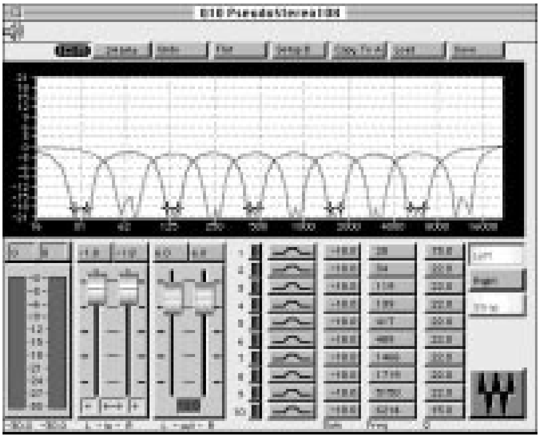
Q10 PShistoric
Q10 PShistoricreverse (not pictured)

A 10-band pseudostereo effect, panning each of 10 bands alternately to the left and to the right of the stereo stage. This setup is not adjustable in frequency and has good mono compatibility. The result is a diffused sound with 'normal' and 'narrow' spread versions. Four bands are reserved for normal EQ effects, preset as hiss and rumble filters, making this an ideal setup for reprocessing historic recordings.

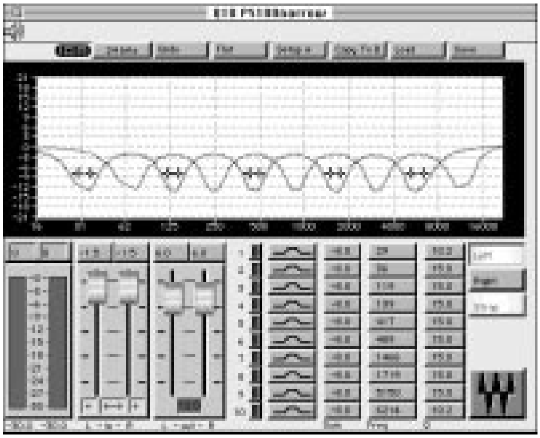


Q10 PS10band
Q10 PS10reverse (not pictured)
Q10 PS10narrow (not pictured)
Q10 PS10revnarrow (not pictured)

A 10-band pseudostereo effect, panning each of 10 bands alternately to the left and right of the stereo stage. Adjustable in frequency, giving a diffused sound, with 'normal' and 'narrow' spread versions. This setup has good mono compatibility.

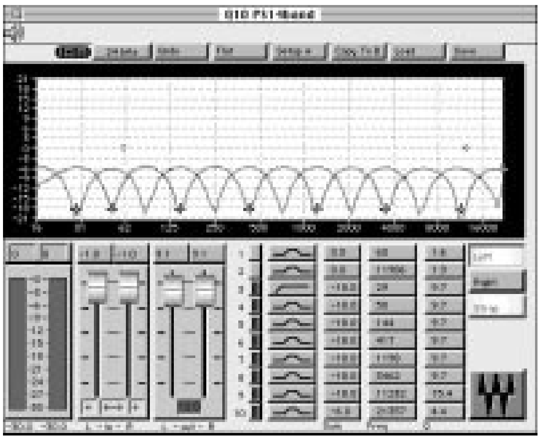


Q10 PseudoStereo10X
Q10 PS10Xreverse (not pictured)



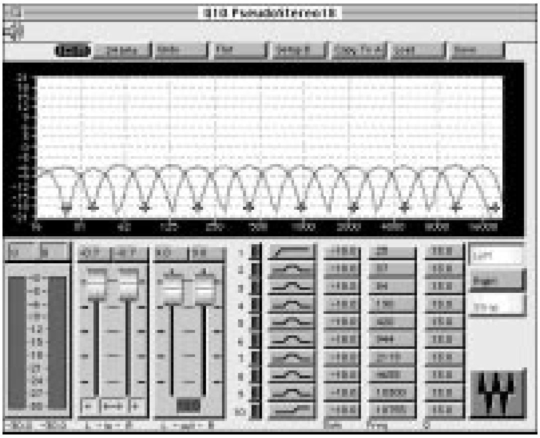
Q10 PS10Xnarrow
Q10 PS10Xrevnarrow (not pictured)

A 10-band pseudostereo effect, panning each of 10 bands alternately to the left and to the right of the stereo stage. This produces a very wide effect at the cost of poor mono-compatibility. This is possibly the most spectacular pseudostereo setup in the library. Not adjustable in frequency.



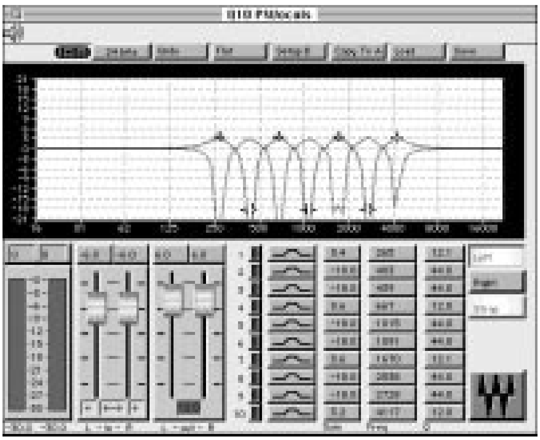
Q10 PS14band
Q10 PS14reverse (not pictured)

A 14-band pseudostereo effect, panning each of 14 bands alternately to the left and to the right of the stereo stage. This gives a diffuse pseudostereo effect, and reasonable mono compatibility, but is not spectacularly wide and tends to sound rather more colored than other effects. Not adjustable in frequency. A Baxandall equalizer is also provided.



Q10 PseudoStereo18
Q10 PS18reverse (not pictured)

An 18-band pseudostereo effect, panning each of 18 bands alternately to the left and to the right of the stereo stage. This gives the most diffuse pseudostereo effect, and reasonable mono compatibility, but is not spectacularly wide and again tends to sound rather more colored than other effects.



Q10 PSVocals
An adjustable pseudostereo effect where the spread is concentrated in the vocal frequency band.

Variations on these effects

All these pseudostereo effects are provided in a 'normal' and a reversed version, which simply has the equalization on the opposite channel to the normal version, so that the left/right panning effect of different frequencies can be reversed in the stereo stage.

Some of the pseudostereo effects also are also provided with 'narrow' and 'reversed narrow' versions which are essentially the same as the above versions except that the degree of stereo spread effect is reduced, for applications where a more subtle effect is preferred. Narrow setups are particularly recommended for speech where the normal spread generally sounds unnaturally wide.

All ten Q10 bands are used as a 'bandgroup' in providing the pseudo stereo effect for the 7, 10 and 18 band varieties. The 2-band varieties uses bands 7 to 10 as a bandgroup, the 10X-band variety uses bands 5 to 10 as a bandgroup, and the 14 band variety uses band 3 to 10 as a pseudostereo bandgroup.

Varying all frequencies together

Be sure to ensure that the two stereo channels are strapped when the any adjustments are made to pseudostereo effects. If you make a mistake and make an irretrievable change to the parameters, simply load the setup again and start over.

All pseudostereo effects, with the exception of the historic, 10X, 14 and 18 band cases, may have all bands moved up and down together. The procedure is to select and drag all the 'frequency' value windows only of the bandgroup (i.e. all bands in the tunable, 7-, and 10-band designs, and bands 7-10 in the 2-band designs). In this way, the operating frequencies of the pseudostereo effect may be moved up or down, This gives a change in the pseudostereo effect without altering its overall tonal balance. This often allows fine tuning to match the effect to particular program material, so that the stereo stage and the positions of frequency components within it are well balanced.

Separate Control of Pseudostereo Bands

Additionally, the tunable, 2-, 7- and 10-band pseudostereo setups allow separate control of individual pseudostereo frequency bands. The details differ in these three cases, and so are described individually.

2-Band Cases. (Q10 PseudoStereo2, Q10 PS2reverse, Q10 PS2band2and Q10 PS2reverse2).

Ensure that the two stereo channels are strapped when the following adjustments are made.

Bands 1 to 6 are not used for the pseudostereo effect so may be used for additional conventional equalization. As configured in the setup file, this provides a 6-band graphic EQ with band centers from 40 Hz to 12 kHz, and band gains may be adjusted up or down from the provided 0 dB flat setting.

The pseudostereo in this case is provided by Q10 bands 7 to 10. To switch the pseudostereo effect in or out, select and click the 4 In/Out buttons of Q10 bands 7 to 10.

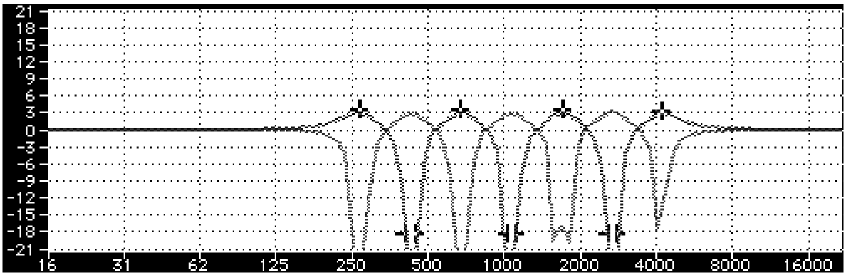
This pseudostereo effect permits independent adjustment of the frequency at which bass signals are panned to one side and at which treble signals are panned to the other. To adjust the bass frequency, select and drag the *frequency* value windows of Q10 bands 7 and 8 as desired. To adjust the treble frequency, select and drag the *frequency* value windows of Q10 bands 9 and 10.

Be aware, however, that the pseudostereo effect will remain tonally neutral only so long as the bass and the treble bands do not overlap too much. The graphic display will show you this. Correct pseudostereo operation will occur if and only if the two curves cross over at about a 0 dB level. If the cross-over point level is significantly below 0 dB, the two bands overlap too much and the tonal quality will be compromised.

The two types of 2-band effect differ in the steepness of crossover between left, center and right positions, with the 'band2' effect giving a less rapid crossover.

7-band case. (Q10 PS7band, Q10 PS7reverse, Q10 PS7band2, Q10 PS7reverse2, Q10 PSVocals).

This setup is recommended where the maximum width of stereo effect is desired, and where mono compatibility is not of major importance. Like the 10-bandX effect, this pseudostereo effect can sometimes show more undesirable side effects on some material than many of the others, so the subjective result should be monitored carefully.



Frequency response of Q10 PSVocals

The '7band2' effect differs from the '7band' effect only in that the frequencies chosen for left and right panning are rather more widely spaced apart. By contrast, the PSVocals case has the bands packed together over a narrow vocal frequency range.

Once again, ensure that the two stereo channels are strapped when the following adjustments are made.

In this case, all ten Q10 bands are used for the pseudostereo effect. To switch the pseudostereo effect in or out, select the In/Out buttons of all ten Q10 bands, and then click anywhere within any of these buttons.

The frequencies of the pseudostereo effect may be adjusted up or down by select-and-dragging all ten frequency value windows in Q10 bands 1 to 10.

The 7-band versions may also be used as the basis for a more flexible adjustment of pseudostereo effect. The effect is divided into 4 bandgroups: bands 1-2, band 4-6, bands 7 -9 and band 10. Any number of these 4 bandgroups may be switched in or out separately, and their frequencies individually adjusted by selecting and dragging all frequency value windows within a bandgroup, provided only that the bandgroups do not overlap too much (i.e. provided that the crossover of left and right graph curves does not fall significantly below 0 dB).

10-Band Cases. (Q10 PS10band, Q10 PS10narrow, Q10 PS10reverse, Q10 PS10revnarrow).

This 10-band pseudostereo effect is probably the most subtle and diffuse of the available setup types. It features a wide flexibility of adjustment and is recommended for when the frequency components simply require a broad spread and do not need to have pronounced stereo positioning.

Ensure that the two stereo channels are strapped when the following adjustments are made.

The pseudostereo in this case is provided by all ten Q10 bands. To switch the pseudostereo effect in or out, select and click all ten In/Out buttons of Q10 bands 1 to 10,.

The frequencies of the pseudostereo effect may be adjusted up or down by select-and-dragging all ten frequency value windows in Q10 bands 1 to 10.

In these setups, the spreading effect may be confined to limited parts of the frequency range only by switching out one or more bands. Any number of adjacent bands may be switched out at either end of the frequency range. In the middle of the frequency range, an even number of adjacent bands may be switched out. Remaining adjacent groups of switched in bands may be varied in frequency by selecting and dragging their frequency value windows. Tonal neutrality will be retained provided that the bands do not overlap too much (i.e. provided that the crossover of left and right graph curves does not fall significantly below 0 dB).

Tunable Pseudostereo Setups

Q10 PStuneableA
Q10 PStuneableAreverse
Q10 PStuneableB
Q10 PStuneableBreverse
Q10 PStuneableC
Q10 PStuneableCreverse

These pseudostereo setups allow tuning of particular frequency bands to the left side or the right side of the image, and are particularly effective for placing elements of the original mix at different positions within the stereo spread. These setups provide the means of doing this providing the sounds being processed are separated enough in frequency.

The four bandgroups for these presents are:

Q10 bands 1 & 2 : narrow frequency band panned to left (right for 'reverse' setups)
Q10 bands 3 to 5 : wide frequency band panned to left (right for 'reverse' setups)
Q10 bands 6 to 8 : wide frequency band panned to right (left for 'reverse' setups)
Q10 bands 9 & 10 : narrow frequency band panned to right (left for 'reverse' setups)

Any one or more bandgroups may be adjusted in frequency by selecting the frequency value windows of that bandgroup(s), and dragging to either side.

The bandgroups may be switched in or out individually, provided that no two 'in' bandgroups overlap too much in frequency range. Excessive overlap can be seen from the graphical display by the crossover of left and right curves for adjacent bands differing (by more than say a dB) from 0 dB. It is normally a good idea for two adjacent panned bands to be to opposite sides of the stereo stage - since this allows much more overlap of bands than the case where they are panned the same way.

The A, B and C versions of the setups differ only in the chosen initial frequencies and in which bandgroups are initially switched in. Since, apart from the initial center frequencies, their bandgroups are identically set up, they can all be used with frequency tuning to achieve identical effects, but it is easiest to choose the version closest to what is required as a starting point.

The A versions of the tunable pseudostereo setups are ideal for achieving a 2-band pseudostereo effect, where one wishes to be able to select using a narrow or wide frequency band after tuning the pseudostereo panning in frequency. In the A version, only two bandgroups are switched in when loaded - the wide frequency bands. By selecting all 10 In/Out buttons and switching, the wide bands are switched out and narrow bands tuned to the <u>same</u> frequencies are switched in. The idea is to tune the frequencies of bands 1 to 5 <u>together</u> by selecting all their frequency value windows and dragging. Similarly, tune the frequencies of bands 6 to 10 <u>together</u> by selecting all their frequency value windows and dragging.

Having chosen the frequencies at which the desired sounds are panned to left and right, you can then choose whether to have narrow or wide frequency range for the panning by selecting the In/Out buttons of bands 1 to 5, and clicking on one of them. This will alternate between a narrow and a wide band at the same frequency. The same procedure can be used independently to select a narrow or wide frequency range for the panning in bands 6 to 10.

The B and C versions of the tunable pseudostereo setups are intended as a starting point for a 4-band tunable pseudostereo effect. They all have the same frequency bands panned to the same positions when loaded. The only difference is the order of narrow and wide bands.

In the B versions, the lowest and highest bands have narrow frequency range, and the mid-low and mid-high bands have wide frequency range. In the C versions, the lowest and highest bands have wide frequency range, and the mid-low and mid-high bands have narrow frequency range.

Inflexible Pseudostereo Setups

Q10 PseudoStereo10X
Q10 PS10Xreverse
Q10 PS10Xnarrow
Q10 PS10Xrevnarro
Q10 PShistoric
Q10 PShistoricreverse
Q10 PS14band
Q10 PS14reverse
Q10 PseudoStereo18
Q10 PS18reverse

These pseudostereo setups are inflexible in offering neither frequency adjustment nor spread adjustment, but in compensation offer a remarkably large number of pseudostereo bands, respectively, 10, 10, 14 and 18, for a much more diffuse effect than setups with a small number of bands. Because there are more bands, there's less likelihood of individual sounds having different frequency components 'split' between different stereo positions.

The '10X' pseudostereo setups provide a 10 band pseudostereo with a very wide stereo stage for when spectacular spread is desired without excessive phasiness. However, mono compatibility is not particularly good.

The 18 band setups provide a very diffuse sound, but not the widest stage, but have fairly good mono compatibility making them suitable for use in areas such as broadcast when a general-purpose pseudostereo effect is required.

The 14-band setups are similar to the 18-band case, except that they only have 14 pseudostereo bands. However, the pseudostereo algorithm now uses as bandgroup Q10 bands 3 to 10, which leaves Q10 bands 1 and 2 free for conventional equalization. As loaded, bands 1 and 2 are configured as Baxandall bass and treble controls, and adjustment of their gains provides respective adjustment of bass and treble. This setup is useful in applications where it is desirable to accomplish EQ and pseudostereo processing in one pass.

The 'historic' setups are similar to the 14- and 18-band pseudostereo, except they have only 10 pseudostereo bands, implemented by Q10 bands 5 to 10. This leaves Q10 bands 1 to 4 free for conventional equalization, when the power of a 4-band parametric EQ is needed. As loaded, band 1 is configured as a high-pass filter for filtering out bass rumble, and Q10 bands 2 to 4 form a bandgroup for a tunable psychoacoustic hiss filter. By selecting the frequency value windows of bands 2 to 4 and dragging, the frequency of action of the hiss filter may be varied.

As loaded, this setup is suitable for the treatment of historic mono material, both to achieve a pseudostereo effect, and to filter hiss (or 78 rpm record scratch noise) and rumble in one EQ pass. The hiss filter has been designed for minimum subjective tonal degradation for a given degree of hiss reduction.

HINT. In the historic and 14-band setups, you can see the EQ effect on the graphic display by selecting the bandgroup In/Out buttons of the pseudostereo bandgroup and clicking on one of them. This will remove the pseudostereo and allow you to see the EQ effect. Clicking again will bring in the pseudostereo effect again.

Chapter 2 - Crossover

About crossover

The ‘crossover’ setups are all designed to split incoming mono signals into two sharply separated bands. Typically, one is in the left output channel, the other in the right where the crossover slope between the bands is of the order of 50 dB per octave. The crossover points are at -3dB so that the total energy in the left and the right channel is roughly flat.

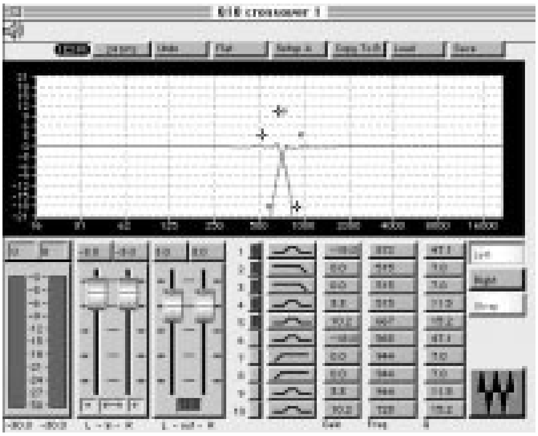
The crossovers are designed so that simply selecting all ten frequency value windows & dragging moves the crossover frequency or frequencies up and down.

These crossover setups can be used to create extreme form of ‘pseudostereo’ where everything in one frequency band is on the left, everything in another frequency band is on the right, and everything in yet another frequency band may be in the middle. With crossover frequencies matched to the musical content, this can place instruments occupying different frequency bands in any of three positions.

The mono compatibility of the crossover effect cannot be guaranteed because of response irregularities in the crossover region due to phase addition and subtraction. However, in cases where the crossover is very sharp, this will only affect a very narrow frequency range, which in practice may be acceptable.

Note that all adjustments of ‘crossover’ effects must use strapped channels. If a file is to be processed, the file should be a mono signal copied as a stereo file, since stereo output files can only be generated from stereo input files.

Q10 crossover1

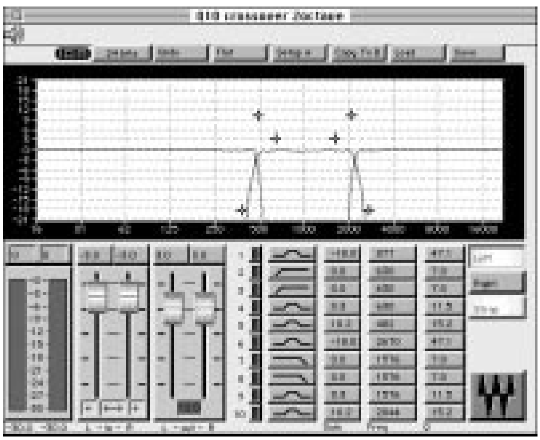


Q10 crossover1

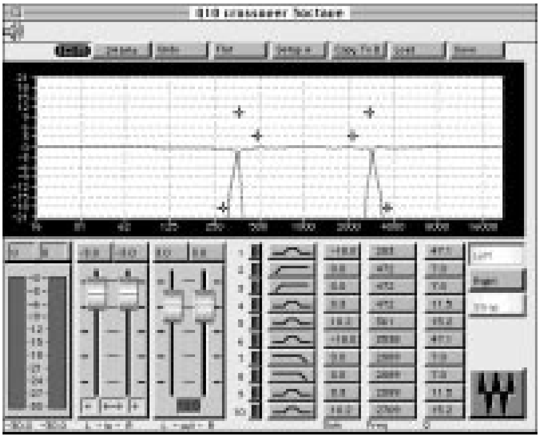
This setup simply splits the incoming signal placing a brickwall low-pass filter in the left channel and a brickwall high-pass filter in the right channel. This setup uses as bandgroup Q10 bands 1-5, placing bass on the left and treble on the right. As loaded, the crossover frequency is 700 Hz, but it may be select and dragged to any other frequency up to several kHz.

Bandgroup Q10 channels 6 to 10 provides an identical crossover except that treble is sent to the right and bass to the left. One may switch between the two crossovers by selecting all ten In/Out buttons and clicking on one of them.

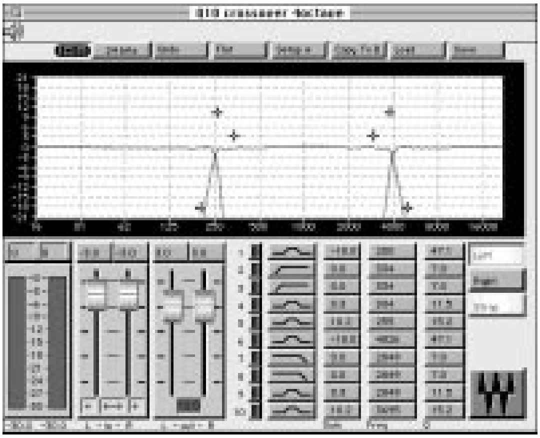
Q10 Crossover 2, 3, and 4



Q10 crossover2octave



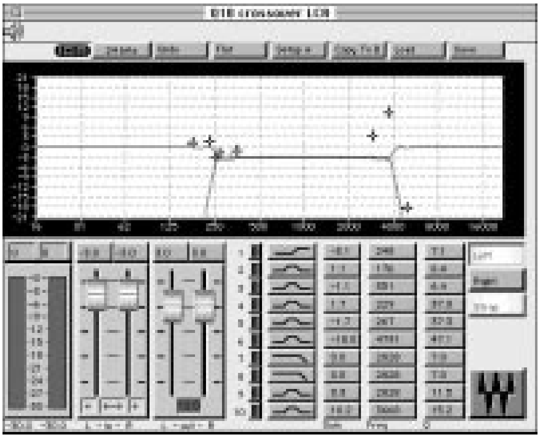
Q10 crossover3octave



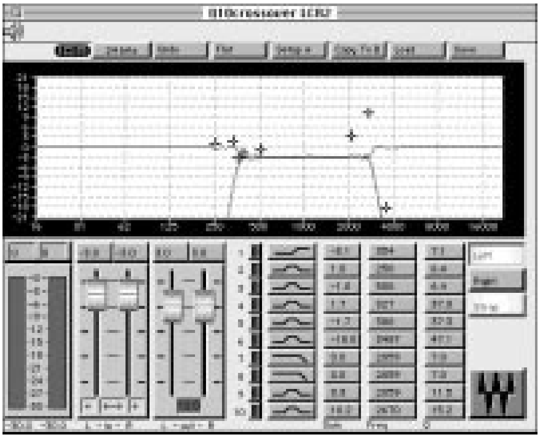
Q10 crossover4octave

These use all 10 Q10 bands as a bandgroup, and put low and high frequencies on the right channel and middle frequencies on the left. The two crossover frequencies are about 2, 3 and 4 octaves apart in the respective setups, with the middle band centered at 1 kHz. The overall frequency may be moved up or down by selecting all ten frequencies and dragging.

A practical example is to use, say, the crossover4octave case to put the vocals of a mono recording on the left, and the bass line and cymbals on the right.



Q10 crossoverLCR



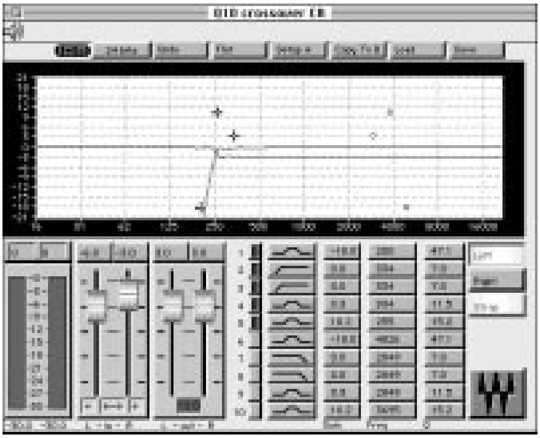
Q10 crossoverLCR2

These setups put low frequencies below a first low crossover frequency on the left, high frequencies above a second high crossover frequency on the right, and everything between the two crossover frequencies in the center at equal levels in both channels. The LCR setup has the spacing between crossover frequencies of 4 octaves, and the LCR2 setup uses 3 octave spacing.

The response of each channel steps down 3 dB over the middle band to ensure that the total energy of the 2 channels remains constant.

The first low crossover frequency is controlled by bandgroup Q10 bands 1 to 5, and may be adjusted up or down by selecting all five frequency value windows of this bandgroup and dragging them. The second high crossover frequency is controlled by bandgroup Q10 bands 6 to 10, and may be adjusted up or down by selecting all 5 frequency value windows of this bandgroup and dragging them.
Please note that at middle frequencies, the phase response on the two channels is not identical, so that there will be some ‘phasiness’ and imperfect mono compatibility.

CR/LC-types



Q10 crossoverCR
Q10 crossoverLC (not pictured)

These two setups are identical except that the channels are the opposite way round.

The LC setup when opened has bandgroup Q10 bands 1 to 5 in, and bandgroup Q10 bands 6 to 10 out. It has the effect of putting frequencies below a first crossover frequency on the left, and frequencies above that crossover frequency in the center. The crossover frequency may be moved up or down by selecting the 5 frequency value windows of Q10 bands 1 to 5 and dragging.

If all 10 In/Out buttons are selected and clicked, so that Q10 bands 6 to 10 are in and bands 1 to 5 are out, then the LC setup has the effect of putting frequencies above a second crossover frequency on the left, and frequencies below that crossover frequency in the center. The crossover frequency may be moved up or down by selecting the 5 frequency value windows of Q10 bands 6 to 10 and dragging.

The CR setup is identical except that panning to the left is replaced by panning to the right throughout.

Other uses of crossover setups

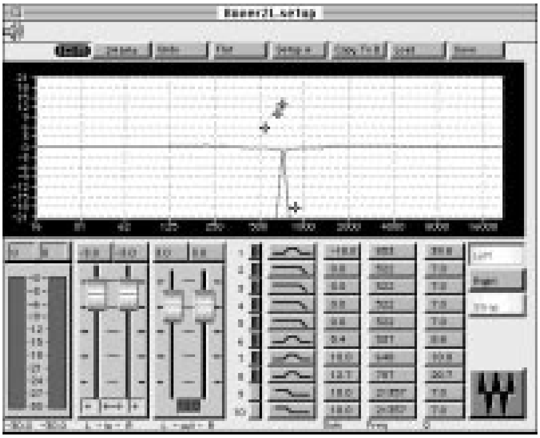
The Crossover setups may also be used as conventional active crossovers for use in the design of loudspeaker systems or in any other test situation where a versatile, precision crossover is required. When used in TDM systems or within WaveShell, signals may be processed in real time without first having to be stored as soundfiles.

Rapid crossover types

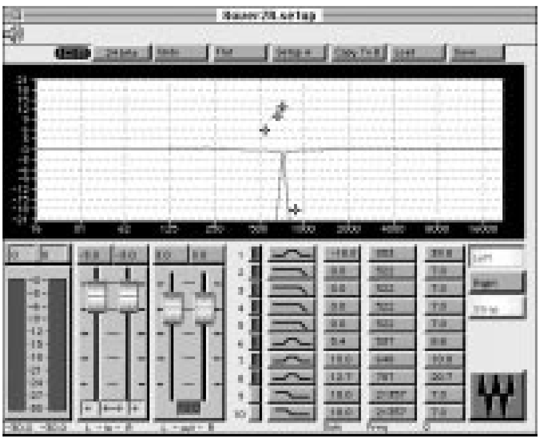
The following "crossover" setups for the Q10 split the audio band into several bands and feed them to left and right channels. They are characterized by flat responses within each frequency range and very rapid crossovers between frequency bands.

Q10 Xover2L
Q10 Xover2R
Q10 Xover2Low
Q10 Xover2Hi

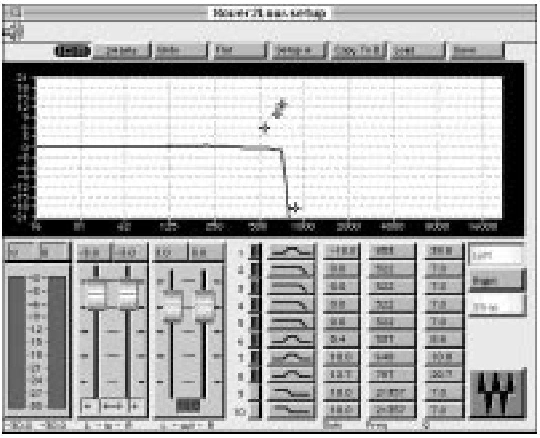
An ultra-rapid crossover bandsplit into 2 bands where the crossover rate is around 150 dB per octave down to about -40 dB. These setups use bandgroup 1-8, and the cross-over frequency may be adjusted (up to a few kHz) by selecting all 8 frequencies in both channels, and dragging up or down.



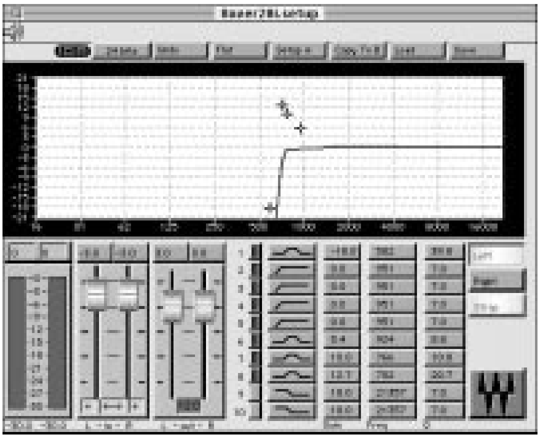
Q10 Xover2L ***puts the low-pass filter in the right channel and the high-pass in the left.***



Q10 Xover2R ***puts the low-pass filter in the left channel and the high-pass in the right.***



Q10 Xover2Low ***puts the low-pass filter in both channels.***



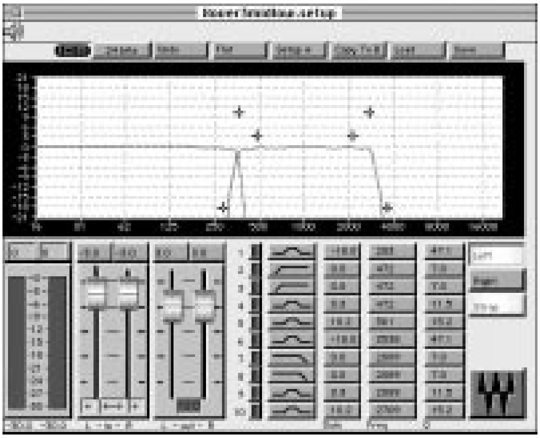
Q10 Xover2Hi ***puts the high-pass filter in both channels.***

The use of these setups allows an effectively clean split of any audio signal into two separate frequency bands, which can be separately processed before being reassembled. Only in the narrow crossover frequency region will there be any notable side effects and interaction between bands.

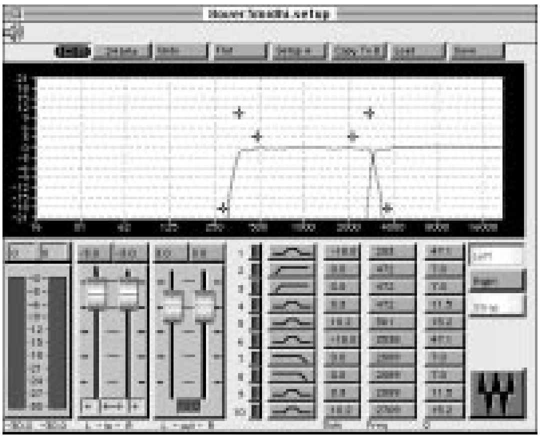
Q10 Xover3midlow
Q10 Xover3midhi
Q10 Xover3low
Q10 Xover3mid
Q10 Xover3hi

A rapid crossover bandsplit into 3 bands where the crossover rate for the low and high pass is around 150 dB per octave down to about -40 dB, and where the mid-passs has rather gentler cross-over rate. The low and high-pass filters use bandgroup 1-8, and the mid-pass filter uses bandgroup 1-5 for its high-pass action and bandgroup 6-10 for its low-pass action.

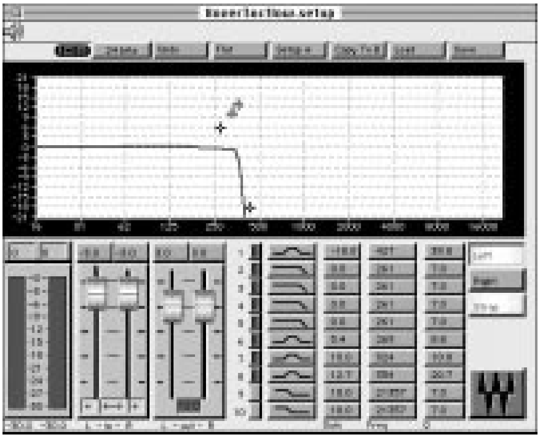
As provided, the mid-pass band covers about 4 octaves, and the cross-over frequencies may be adjusted (up to a few kHz) by selecting all 10 frequencies on both channels and dragging up or down.



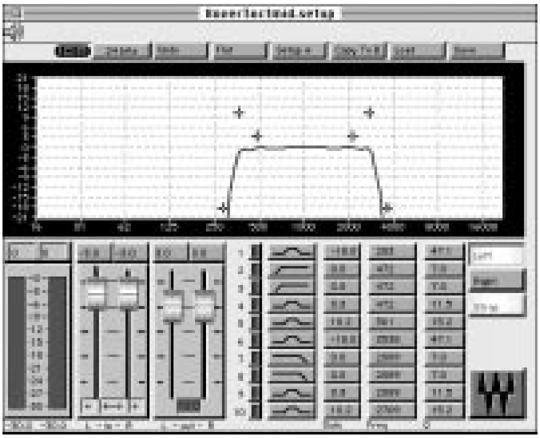
Q10 Xover3midlow ***puts the low-pass filter in the right channel and the mid-pass in the left.***



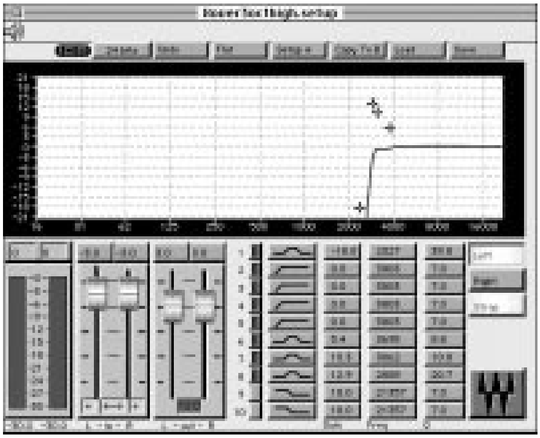
Q10 Xover3midhi ***puts the mid-pass filter in the left channel and the high-pass in the right.***



Q10 Xover3octlow **puts the low-pass filter in both channels.**



Q10 Xover3octmid **puts the mid-pass filter in both channels.**



Q10 Xover3octhi ***puts the high-pass filter in both channels.***

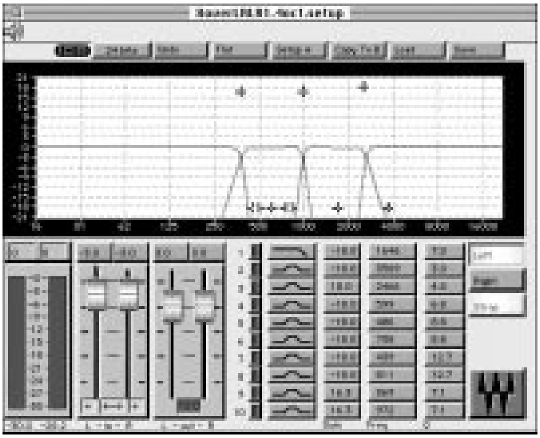
The use of these setups allows a reasonably clean split of any audio signal into three separate frequency bands, which can be separately processed before being reassembled. Only in the crossover frequency regions will there be any notable side effects and interaction between bands.

LRLR types

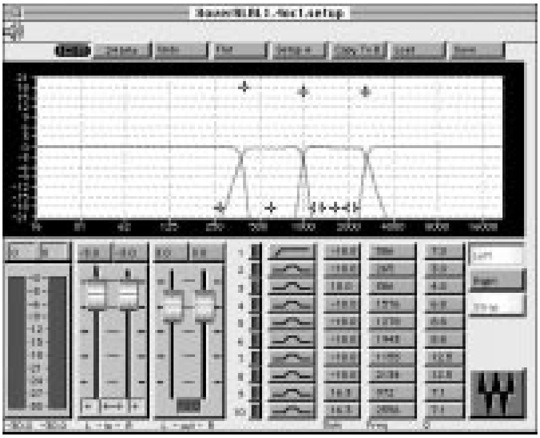
Q10 XoverLRLR1.4oct
Q10 XoverRLRL1.4oct
Q10 XoverLRLR1.9oct
Q10 XoverRLRL1.9oct
Q10 XoverLRLR2.4oct
Q10 XoverRLRL2.4oct

A rapid crossover bandsplit into 4 bands where the rejection out of band exceeds -35 dB in the low-mid and high-mid bands. One channel passes the low and high-mid bands and the other the low-mid and high bands. The width of the low-mid and high-mid bands are equal, and are respectively 1.4, 1.9 and 2.4 octaves depending on the setup chosen.

All 10 bands are used, and the cross-over frequencies may be moved up or down together, up to a few kHz, by selecting and dragging all ten frequencies in both channels.



Q10 XoverLRLR1.4oct



Q10 XoverRLRL1.4oct

The LRLR setups put the low and high-mid filtering in the left channel and the low-mid and high filtering in the right channel. The RLRL setups put the low and high-mid filtering in the right channel and the low-mid and high filtering in the left channel.

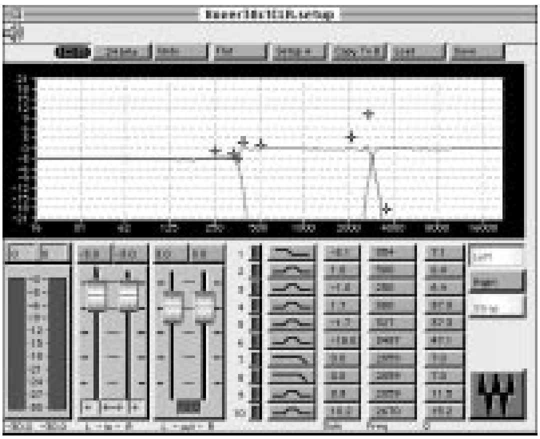
With a dual-mono input, these setups may be used to obtain a fairly extreme pseudostereo effect with a pronounced left-right split.

3-way types

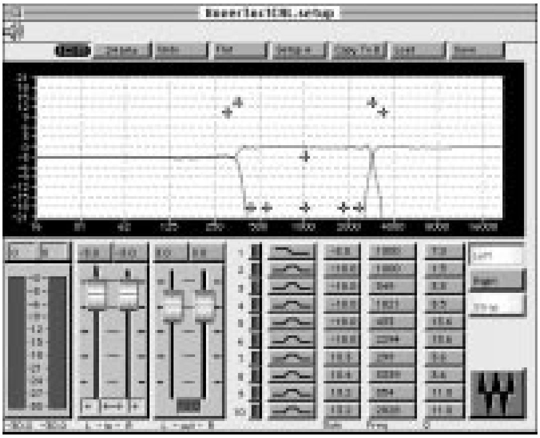
Q10 Xover30ctCLR
Q10 Xover30ctCRL
Q10 Xover3.50ctCLR
Q10 Xover3.50ctCRL
Q10 Xover40ctCLR
Q10 Xover40ctCRL

A rapid crossover bandsplit into 3 bands designed primarily for pseudostereo processing of dual-mono inputs, where the low frequency band is assigned to stereo center, the mid band to one channel and the high band to the other. The separation in the mid band region exceeds 35 dB.

The setups use all 10 bands, and all crossover frequencies may be moved up or down together, up to a few kHz, by selecting and dragging all ten frequencies in both channels. The width of the mid band is 3, 3.5 or 4 octaves depending on the setup chosen.



Q10 Xover30ctCLR

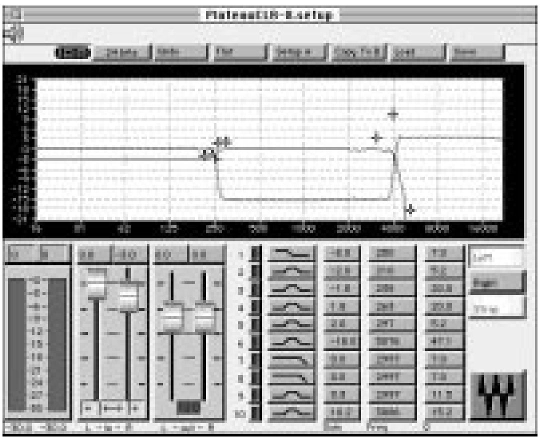


Q10 Xover30ctCRL

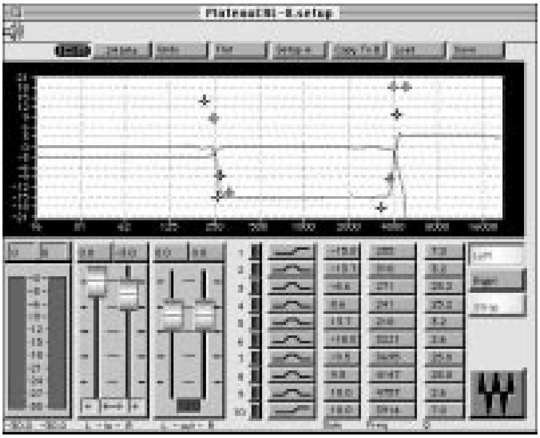
The CLR setups put the mid filtering in the left channel and the high filtering in the right channel. The CRL setups put the mid filtering in the right channel and the high filtering in the left channel.

Q10 PlateauCLR-X
Q10 PlateauCRL-X

These two setups have a similar rapid 3-way pseudostereo bandsplit to the previous setups, but with different performance specifications and adjustability.



Q10 PlateauCLR-X



Q10 PlateauCRL-X

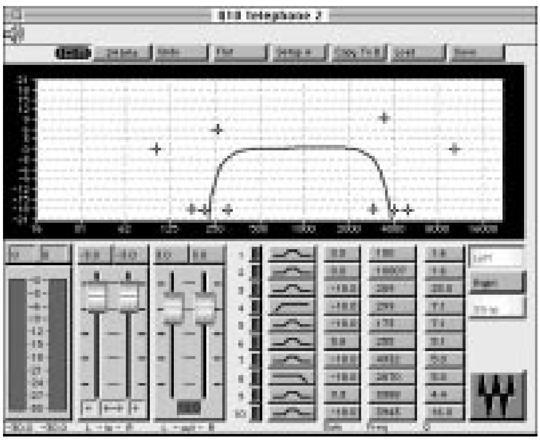
In these setups, the mid-band separation is only 18 dB, which gives a noticably narrowed pseudostereo effect. The high-band separation is still wide.

However, the crossover frequencies may be adjusted independently up to a few kHz. Bandgroup 1-5 controls the lower cross-over frequency and bandgroup 6-10 the higher crossover frequency. A crossover frequency may be moved up or down, up to a few kHz, by selecting and dragging all five frequencies in both channels in the selected bandgroup.

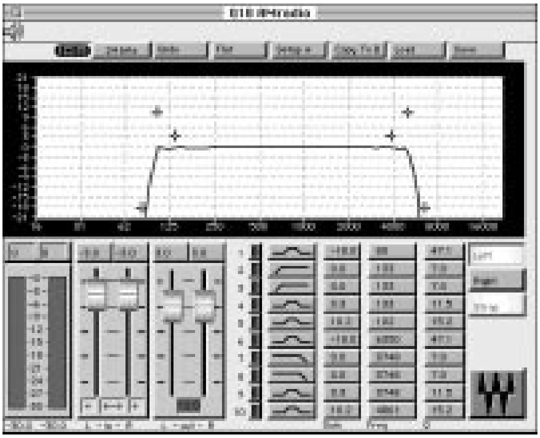
Chapter 3 - Bandlimiting

Many audio signals are destined to be heard via systems or transmission channels that impose a quality limitation on the material being transmitted, and it is often helpful to know what audible effects such a system will have on a given piece of program material. For example, if important parts of a mix become inaudible or out of balance after being passed through a Q10 emulation of the transmission system, corrective action (such as preparing a specially EQ'd or balanced mix) may be taken. Three of the setups provided are particularly useful in this type of application and are very similar except for the precise choices of cut-off frequencies in the bass and treble:

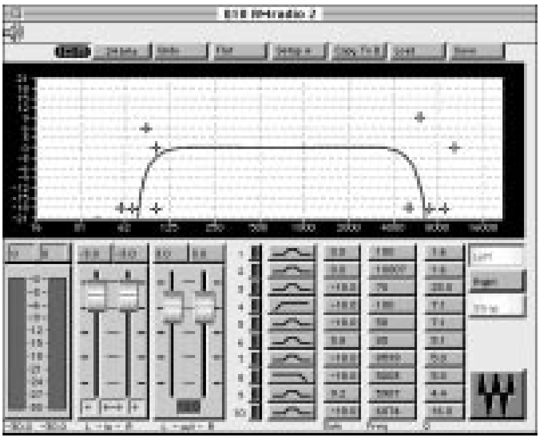
Q10 telephone: simulates a typical telephone bandwidth of 300 Hz to 3 kHz, using a very steep 'brickwall' cut-off outside this band. Unlike many typical telephones, however, it is flat within this band. This setup is useful both for determining the effect of program material likely to be heard down telephone lines and for simulating a 'telephone' effect when needed for drama, commercials or an effect in music.



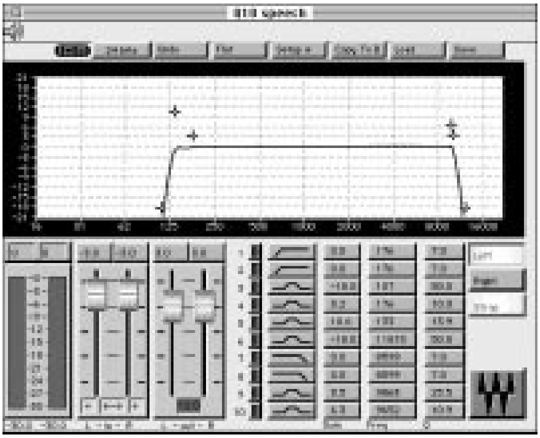
Q10 telephone2: also simulates a typical telephone bandwidth of 300 Hz to 3 kHz, using a rather less steep cut-off outside this band. Again the response is flat within this band. Q10 bands 1 and 2 allow extra equalization to be provided but are initially set to flat. If their gains are altered, they provide Baxandall bass and treble controls.



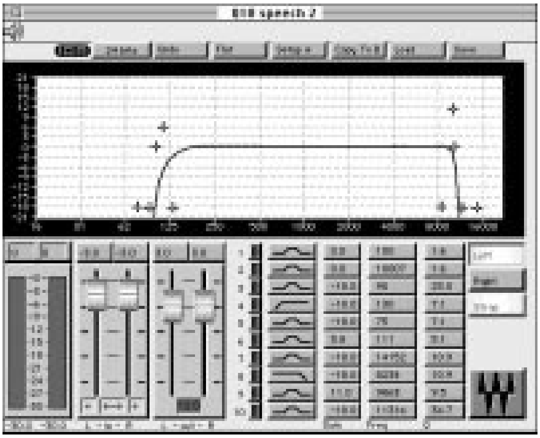
Q10 AMradio: simulates a typical received AM radio bandwidth of 100 Hz to 5 kHz, using a very steep cut-off outside this band. Unlike many typical AM radios, however, it is flat within this band. **Q10 AMradio** provides the steepest brickwall filtering for the best effect. This setup is useful both for determining the effect of program material likely to be heard on AM radios and for simulating a ‘radio’ effect when needed for drama, commercials or as an effect in music.



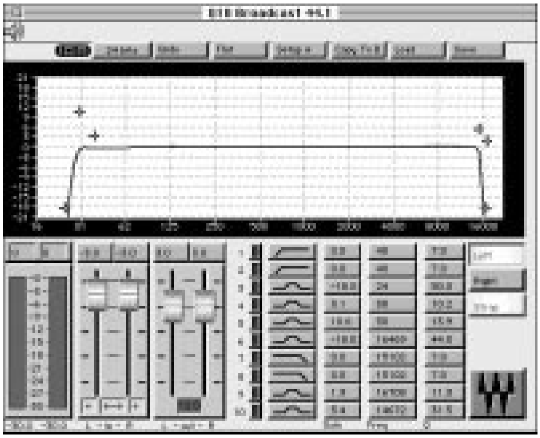
Q10 AMradio2: provides a less sharp cutoff simulation of the AM radio bandwidth 100 Hz to 5 kHz. Q10 bands 1 and 2 allow extra equalization to be provided but are set to flat as provided. If their gains are altered, they provide Baxandall bass and treble controls.



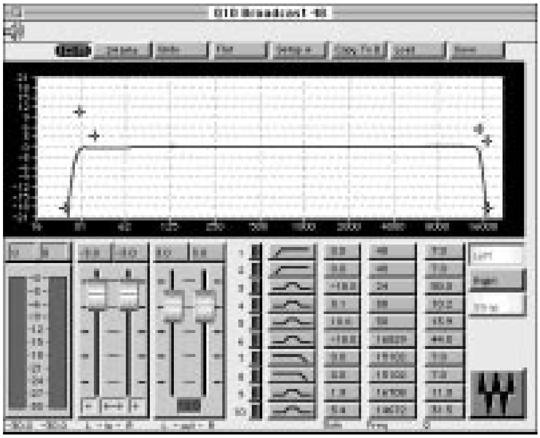
Q10 speech: provides very steep 'brickwall' cut-offs at 130 Hz and 10 kHz, and a flat response in between. It is designed for limiting the bandwidth of speech in applications such as multimedia or some broadcast applications where an extended speech bandwidth may be undesirable. Typically, this setup may be used when speech intelligibility and clarity is more important than true fidelity, or where the loudspeaker system on which the material will be reproduced has a limited bass response.



Q10 speech2: provides slightly less sharp cut-offs at 130 Hz and 10 kHz, and a flat response in between. Q10 bands 1 and 2 allow extra equalization to be provided and are set to flat as provided. If their gains are altered, they provide Baxandall bass and treble controls.



Q10 Broadcast44.1



Q10 Broadcast48

Q10 Broadcast44.1, Q10 Broadcast48. permits files recorded at the respective 44.1 and 48 kHz sampling rates to be bandlimiting to from 30 Hz to 15 kHz. This is the frequency range used for FM broadcasting and TV work. This setup can be used as an antialiasing filter prior to sampling rate conversion to the widely used broadcast sampling rate of 32 kHz. Not adjustable.

Adjusting these setups

Q10 telephone, Q10 AMradio, Q10 speech

These three setups all allocate the Q10 bandgroup bands 1 to 5 for the high-pass (low cut) filter, and the Q10 bandgroup bands 6 to 10 for the low-pass (treble cut) filter.

The high-pass filter and low-pass filter may be used individually or together.

High-pass filter. This uses the bandgroup Q10 bands 1 to 5. To switch the high-pass in or out, select thpcì¨yn/Out buttons of Q10 bands 1 to 5, and click on any one of these buttons. To adjust the frequency of cutoff of the high-pass filter up or down, select the 5 frequency value windows of Q10 bands 1 to 5, and click and drag the frequency on any one of them.

Low-pass filter. This uses the bandgroup Q10 bands 6 to 10. To switch the low-pass in or out, select the 5 In/Out buttons of Q10 bands 6 to 10, and click on any one of these buttons. To adjust the frequency of cutoff of the high-pass filter up or down, select the 5 frequency value windows of Q10 bands 6 to 10, and click and drag the frequency on any one of them.

Note from the graphical display that the frequency range over which the brickwall low-pass remains reasonably flat varies with the cut-off frequency setting, and for best results it is advisable to start with the setup with the low-pass cut-off frequency closest to the one desired.

Q10 telephone2, Q10 AMradio2, Q10 speech2

These three setups all allocate Q10 bands 1 and 2 to the respective bass and treble of a Baxandall tone control, and allocate the Q10 bandgroup bands 3 to 6 for the high-pass (low cut) filter, and the Q10 bandgroup bands 7 to 10 for the low-pass (treble cut) filter.

The high-pass filter and low-pass filter may be used individually or together.

High-pass filter. This uses the bandgroup Q10 bands 3 to 6. To switch the high-pass in or out, select the four In/Out buttons of Q10 bands 3 to 6, and click on any one of these buttons. To adjust the frequency of cutoff of the high-pass filter up or down, select the four frequency value windows of Q10 bands 3 to 6, then click and drag the frequency on any one of them.

Low-pass filter. This uses the bandgroup Q10 bands 7 to 10. To switch the low-pass in or out, select the four In/Out buttons of Q10 bands 7 to 10, and click on any one of these buttons. To adjust the frequency of cutoff of the high-pass filter up or down, select the four frequency value windows *and the four Q value windows* of Q10 bands 7 to 10, then click and drag the frequency and Q together on any one of them. Dragging frequency alone will not give a flat response before cut-off as the frequency is varied. In practice, the low-pass filter will work well for '-3 dB' cut-off frequencies in the range 2 to 13 kHz approximately.

Alternative low- and high-pass filters.

Alternative low- and high-pass filters with a slightly gentler cut off rate are provided in **Q10 EQtools**. In this instance, the high pass filter uses the bandgroup of Q10 bands 5 to 7, and may be switched in and out by selecting and clicking the three buttons of this bandgroup, and the cut-off frequency may be varied by selecting and dragging the three frequency value windows of this bandgroup.

In this case, the low-pass filter uses the bandgroup of Q10 bands 8 to 10, and may be switched in and out by selecting and clicking the three buttons of this bandgroup, and the cut-off frequency may be varied by selecting and dragging the three frequency value windows *and the three Q value windows* of this bandgroup. For the low-pass filter, dragging frequency alone will not give a flat response before cut-off as the frequency is varied.

Hiss filtering. The above low-pass filters can be used to reduce hiss on hissy material, but the user is cautioned that this is generally not the most effective form of filtering for reducing audible hiss with minimum side effect. This is because the ears are most sensitive to the frequency region around 4 kHz, with a low sensitivity to very high frequencies. A low-pass filter removes all the very high frequencies, despite the fact that the ears are less sensitive to hiss at these highest frequencies. A better strategy is to use filters designed to reduce or remove only those frequencies at which hiss is most audible, and to leave lower *and higher* frequencies alone. Such filters are provided in other setups as described below.

Also, very steep-cut, low-pass filters are found to give the remaining hiss an unpleasant 'bright' and edgy quality that is sometimes more objectionable than leaving hiss in. The use of low-pass filters to remove hiss does not generally produce the best subjective results.

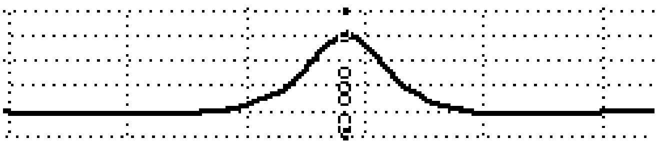
Chapter 4 - EQ Tools

Superparametric EQ

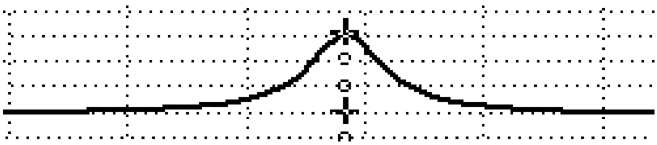
Parametric equalizers, while extremely useful, rarely provide exactly the subjective effect required by audio production engineers. Their action might more appropriately be described as approximate, because the ears are surprisingly sensitive to small variations in frequency response shape, Even relatively small boosts or cuts away from the desired frequencies can be unpleasantly audible.

Conventional parametric ‘bell’ boosts and cuts not only boost and cut the desired frequency band, but they also affect frequencies well away from the target frequencies causing unpleasant coloration.

Several setups provided here provide ‘Superparametric’ boosts and cuts that are more precise in their action, affecting only the desired frequency range and having virtually no effect outside from that range. When used to cut a desired frequency band, the result is less tonal dulling of other frequencies, and when used to boost a desired frequency band, the result is less tonal alteration of the rest of the sound.



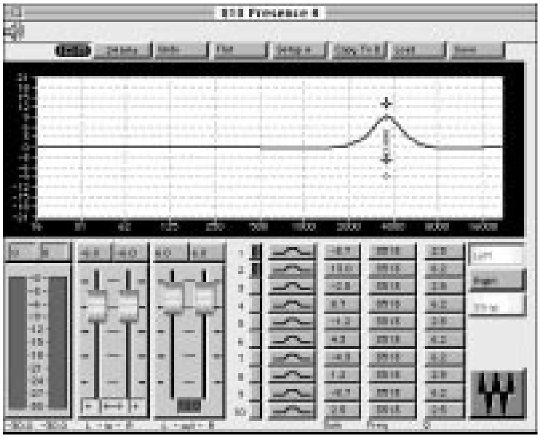
Frequency response of Superparametric boost.



Frequency response of ordinary parametric boost.

To hear the difference between parametric and Superparametric boosts, load **Q10presenceA** in setup A, and load **Q10presenceB** in setup B. **Q10presenceA** is a 9.3 dB Superparametric boost at 3.5 kHz, whereas **Q10presenceB** provides an ordinary 9.3 dB parametric boost at 3.5 kHz, carefully matched in the boost band. Switching between setup A (Superparametric), setupB (parametric) and bypass on speech, it will be heard that the Superparametric EQ is tonally more natural than the parametric.

Adjusting Superparametric EQ

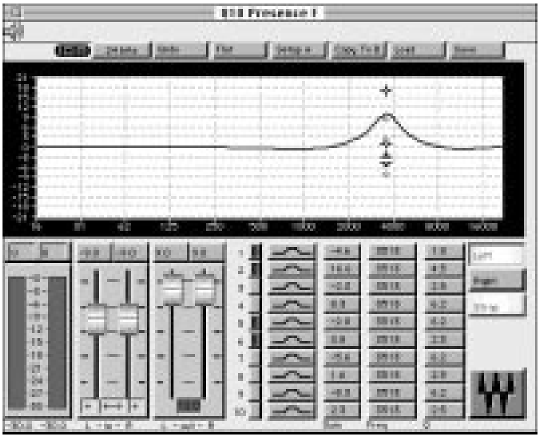


Q10 PresenceA: ***provides five alternative degrees of boost or cut at the presence frequency of 3.5 kHz.***

The bandgroup Q10 bands 1&2 provide +6.2 dB boost at 3.5 kHz.
The bandgroup Q10 bands 3&4 provide +9.3 dB boost at 3.5 kHz.
The bandgroup Q10 bands 5&6 provide +3.1 dB boost at 3.5 kHz.
The bandgroup Q10 bands 7&8 provide 3.1 dB cut at 3.5 kHz.
The bandgroup Q10 bands 9&10 provide 6.2 dB cut at 3.5 kHz.

Select the In/Out button of the desired bandgroup to obtain the desired degree of boost or cut.

To vary the center frequency, select all frequency value windows and drag the frequency to either side as desired. The value windows show the selected frequency. To vary the width of the Superparametric EQ, select all Q value windows and drag the Q to either side as desired.



Q10 PresenceF: A shortcoming of the previous setup is the crude set of values of gain provided - only 6 values (including flat 0 dB) in 3.1 dB steps. If a finer and wider range of values is desired, Q10 PresenceF may be used instead, although selection of gain is much less straightforward. Here also, only a limited range of gains is provided, but other gains may be achieved by switching in two or more bandgroups. The gain may by altered in 2 dB steps between -12 dB and + 12 dB.

Five bandgroups provide five alternative degrees of boost or cut at the presence frequency of 3.5 kHz.

The bandgroup Q10 bands 1&2 provide +12 dB boost.
The bandgroup Q10 bands 3&4 provide +6 dB boost.
The bandgroup Q10 bands 5&6 provide -2 dB cut .
The bandgroup Q10 bands 7&8 provide -4 dB cut.
The bandgroup Q10 bands 9&10 provide -6 dB cut.

To obtain the desired degree of boost or cut, switch in the following bandgroups:

+12 dB, bandgroup 1&2 only.
+10 dB, bandgroups 1&2 and 5&6 only. (10 dB = 12 dB - 2 dB)
+8 dB, bandgroups 1&2 and 7&8 only. (8 dB = 12 dB - 4 dB)
+6 dB, bandgroup 3&4 only.
+4 dB, bandgroups 3&4 and 5&6 only. (4 dB = 6 dB - 2 dB)
+2 dB, bandgroups 3&4 and 7&8 only. (2 dB = 6 dB - 4 dB)
0 dB, all bandgroups out.
-2 dB, bandgroup 5&6 only.
-4 dB, bandgroup 7&8 only.
-6 dB, bandgroup 9&10 only.
-8 dB, bandgroups 5&6 and 9&10 only. (-8 dB = -6 dB - 2 dB)
-10 dB, bandgroups 7&8 and 9&10 only. (-10 dB = -6 dB - 4 dB)
-12 dB, bandgroups 5&6, 7&8 and 9&10 only. (-12 dB = -6 dB - 4dB - 2 dB).

To vary the center frequency, select all frequency value windows and drag the frequency to either side as desired - the value windows show the selected frequency. To vary the width of the Superparametric EQ, select all Q value windows and drag the Q to either side as desired.

Uses of Superparametric EQ

The Superparametric EQ's ability to operate on a selected band of frequencies without affecting neighboring frequencies has a variety of different uses.

One application is to make a sound 'buried' in a mix more audible, by selectively boosting its most prominent frequencies. For example, buried vocals in live recordings can often be 'pulled out' by boosting frequencies between 1.5 or 2 kHz. Using Superparametric EQ, this can be done with less effect on the rest of the mix.

A converse application is to pull down an over-prominent sound in a mix by cutting its most prominent frequencies using a Superparametric cut, again with less likelihood of adversely affecting the rest of the mix.

A more sophisticated use is to improve a mix where several sounds have similar frequency balances causing them to get in each other's way. By selectively boosting some sounds and cutting others in the same frequency band using a Superparametric EQ (ideally before mixdown), it is possible to alter the relative audibility of individual sounds in the mix while having very little subjective effect on the overall sound of the mix.

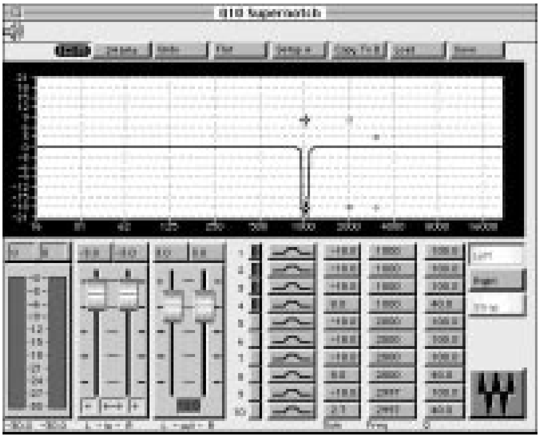
To a limited extent, even applying a Superparametric boost in one stereo channel and a complementary cut on the other can sometimes rescue a marginal mix, if the sounds affected are towards different sides of the stereo stage. However, attempting to overcompensate using this method may lead to a noticeable deterioration in sound quality.

A further important use of Superparametric EQ is for filtering out unwanted noises, including rumble noise, noises due to air conditioning, and hiss. This can be used in much the same manner as parametric EQ, except that the effect on other sounds is lessened. A general hint here is to locate the problem frequency by setting up a Superparametric boost, with fairly narrow width, and to tune it in frequency until the problematic noise is most audible. A Superparametric cut may then be applied at this frequency. It is also helpful to reduce the width of the Superparametric notch so that it is as narrow as possible consistent with the desired degree of reduction of the unwanted noise. The optimum setting is the once which achieves the best compromise between the removal of unwanted noise and audible degradation of the wanted signal.

When a deep notch is required as a noise filter, you can use the Supernotch (Bandgroup Q10 bands 3 & 4) of the **Q10 EQtools**. This may be adjusted in frequency by selecting the two frequency value widows and dragging to either side as required, and may be adjusted in width by selecting the two Q value windows and dragging to either side.

Superparametrics used in this way can make very effective hiss filters. They are generally far more effective than using low-pass filters, however, one warning is in order. Some less good monitoring speakers may well have prominent emphasis frequency bands, and there is a risk with Superparametric dips or notches that you may be tuning out a loudspeaker resonance rather than a true peak in the audible noise. Use of good neutral monitoring loudspeakers will minimize this risk, but it is a wise precaution where possible to check the results of mixes done with Superparametric filters on alternative high quality monitoring situations.

Q10 Supernotch



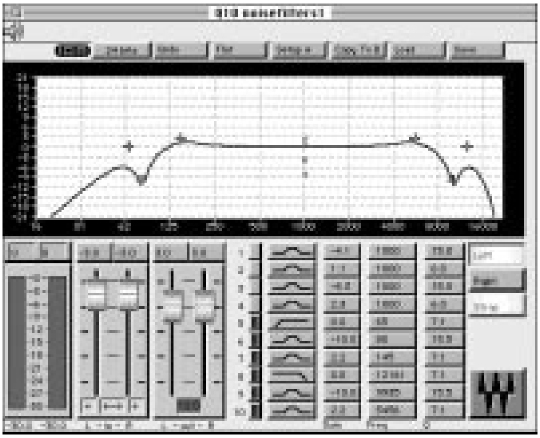
This type of filter is intended to remove whistles or highly pitched interference.

This setup provides a deep narrow tunable notch with about 46 dB attenuation, using as bandgroup Q10 bands 1 to 4. Its frequency may be tuned by selecting the four frequency value windows and dragging. The bandgroup Q10 bands 5 to 8 provides a second similar Supernotch tuned to twice the frequency, which may be switched in if there is a second harmonic present, or tuned to a second independent whistle frequency. The bandgroup Q10 bands 9 & 10 provides a 15 dB attenuation Supernotch set to three times the frequency for the case where a third harmonic component is also present, or where another frequency requiring less attenuation needs to be removed.

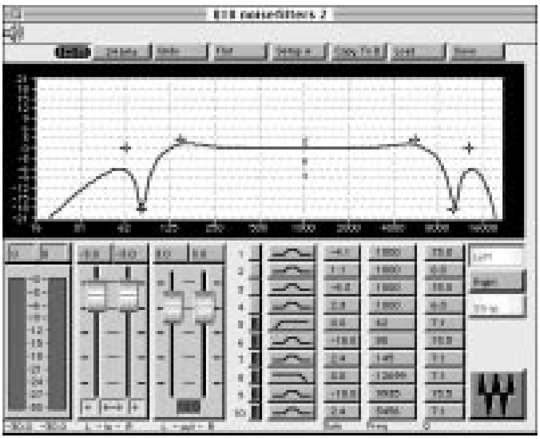
All frequencies may be dragged up and down together if all 10 frequency value windows are selected.

Psychoacoustic noise filters

While Superparametric notches are excellent for reducing noises that are most predominant around a discrete frequency, it is common to encounter material with broadband rumble or hiss noise where Superparametric notches are not adequately effective for reducing noise. In this situation, low-pass filters are needed to remove hiss and high-pass filters to reduce rumble. But psychoacoustically, steep cut filters are far from optimum in this application because they apply an unnecessary degree of filtering to less audible noise components.



Q10 noisefilters1



Q10 noisefilters2

The setups **Q10 noisefilters1** and **Q10 noisefilters2** provide a low-pass and a high-pass filter much better matched to human psychoacoustics. As may be seen from the filter curves on the Q10 graphic display, these filters embody several characteristics found to improve subjective tonal quality:

(i) they rise about 1.5 dB before the cut-off.
(ii) they fall to a deep notch.
(iii) they rise again to a reduced level of about -6dB outside the main band before finally falling away again.

This characteristic is very effective at reducing noise while still leaving enough of the high frequencies present (in the low-pass case) to give a much more natural sound quality than a simple roll-off. The two setups differ only in that the '1' filters have a shallower notch than the '2' filters.

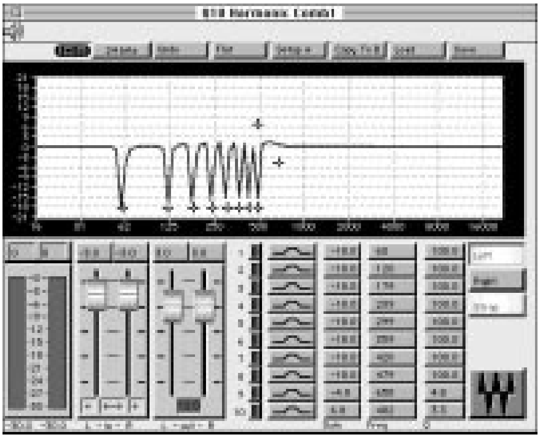
The low-pass hiss noise filter uses bandgroup Q10 bands 8 to 10, and its frequency may be altered by selecting the three frequency value windows and dragging to either side as required, and the low-pass filter may be switched in or out by selecting the three In/Out buttons of bandgroup 8 to 10 and clicking on one of them.

The high-pass bass noise filter uses bandgroup Q10 bands 5 to 7, and its frequency may be altered by selecting the three frequency value windows and dragging to either side as required, and the low-pass filter may be switched in or out by selecting the three In/Out buttons of bandgroup 5 to 17 and clicking on one of them.

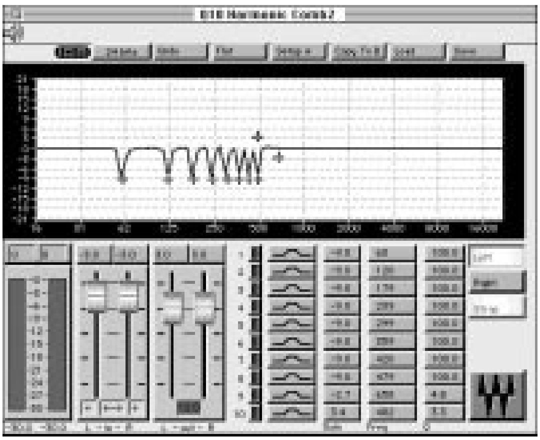
Additionally, two Superparametric dips are provided for additional selective noise filtering in **Q10 noisefilters1** and **Q10 noisefilters2**. One giving a 6 dB dip uses bandgroup Q10 bands 1 & 2, the other giving a 3 dB dip uses bandgroup Q10 bands 3 & 4. They may be tuned individually in frequency by selecting the respective two frequency value windows and dragging to either side, or they may be tuned together by selecting all four frequency value windows if both are used together to provide a 9 dB dip. The width of the Superparametric dip may be altered by selecting two or all four Q value windows and dragging to either side.

Chapter 5 - Harmonic Comb

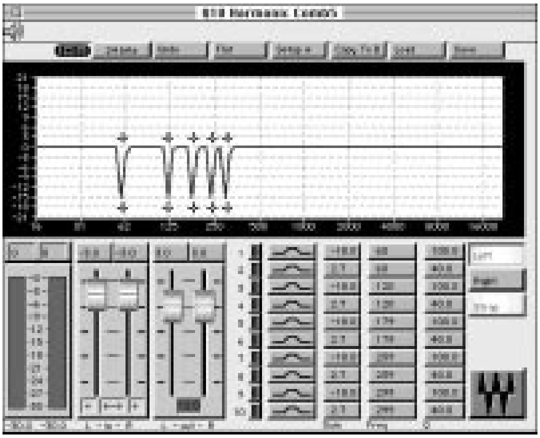
These effects are comb filters designed to notch out the fundamentals and first few harmonics of interference which occurs at a particular frequency.



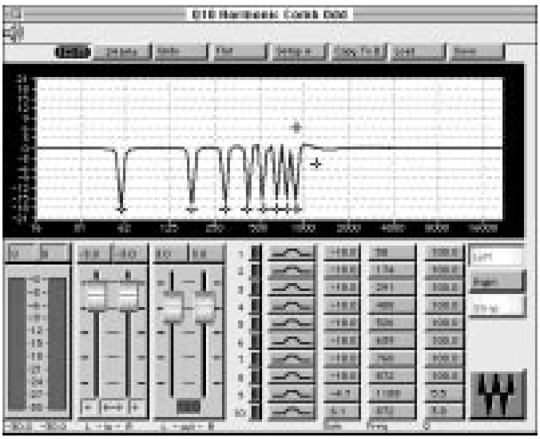
Q10 HarmonicComb1 notches out a fundamental frequency (seen in the top frequency value window) and the 2nd to eighth harmonic. Notch attenuation is around 15 to 18 dB.



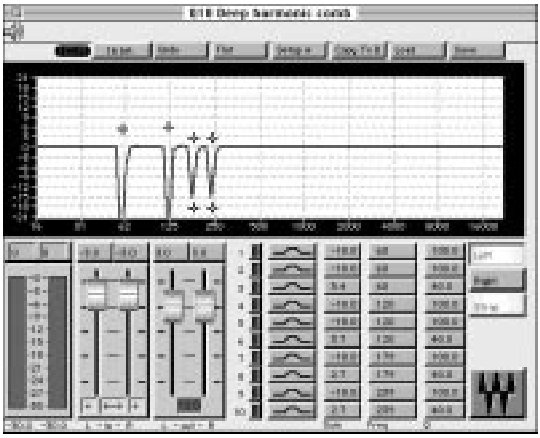
Q10 HarmonicComb2 is the same as **Q10 HarmonicComb1** except that the notch depth is only around 9 dB, for use when less attenuation is required.



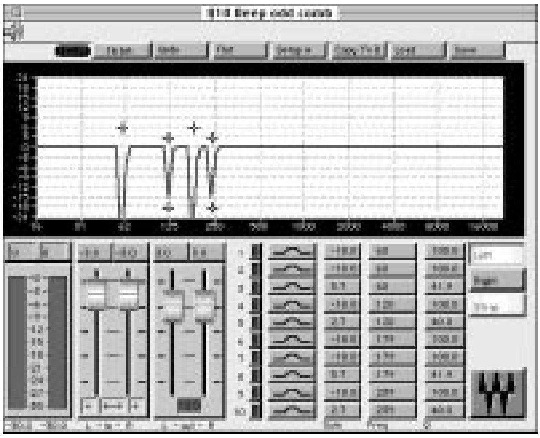
Q10 HarmonicComb5 is similar to **Q10 HarmonicComb1** except that only harmonics up to the fifth are removed, and the notch shape is more precisely controlled and narrower, in order to have less audible effect on wanted signals. The fundamental to fifth harmonic each occupy a bandgroup comprising 2 Q10 bands, respectively 1&2, 3&4, 5&6, 7&8, 9&10 for the fundamental and second to fifth harmonics. Individual notches may be switched out by switching out its bandgroup.



Q10 HarmonicCombOdd filters out the fundamental and odd harmonics only, from third to fifteenth harmonic.



Q10 Deep harmonic comb



Q10 Deep odd comb

These effects are comb filters designed to notch out the fundamentals and first four harmonics of a particular frequency, such as a 60 Hz line power frequency interference or an acoustic "whine" due to high-speed motors.

The **Q10 Deep harmonic comb** setup notches out a fundamental frequency (seen in the top frequency value window) and the 2nd harmonic with a deep notch attenuation of around 30 dB, and the 3rd and 4th harmonics with a lesser notch attenuation in both cases of about 15 dB.

The **Q10 Deep odd comb** setup notches out a fundamental frequency (seen in the top frequency value window) and the 3rd harmonic with a deep notch attenuation of around 30 dB, and the 2nd and 4th harmonics with a lesser notch attenuation in both cases of about 15 dB.

In both cases, the fundamental frequency may be tuned up or down by selecting all 10 frequency value windows and dragging. When the setups are opened, the fundamental frequency is the US power frequency of 60 Hz, useful for attenuating recorded hum components. The setups may be modified to the European 50 Hz power frequency by European users by selecting all frequency value windows and dragging so that all frequency value windows are a multiple of 50 Hz.

These comb filters have some tonal effect on the sound, and are provided as an occasionally useful emergency "sound cleanup" method. They will not work when the pitched interference is a "buzz" with mainly very high harmonics.

Chapter 6 - Multimedia

It is common to do basic production work for multimedia 22 or 11 kHz sampling rates at the CD rate of 44.1 kHz and then to convert the sampling rate. Several tools for improving quality when doing this are provided. The following are bandlimiting (anti-aliasing) setups for files subsequently to be converted to 22 and 11 kHz sampling rates:

Q10 multimedia22
Q10 multimedia11

and these 3 files compensate for defects in the Sound Designer 2 sampling rate conversion process:

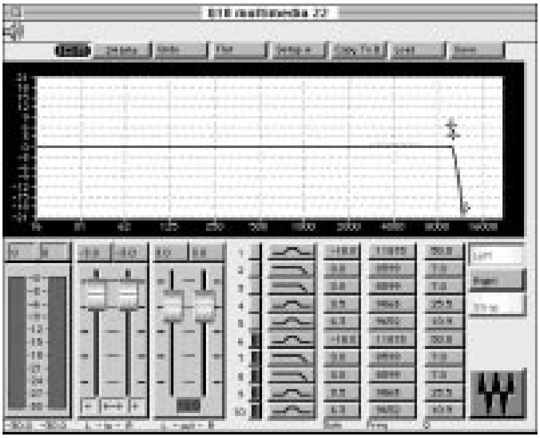
Q10 22.05postcomp
Q10 22.254postcomp
Q10 11.025postcomp

Note: These files are ONLY for use with Sound Designer II version 2.7 and below. Beginning with Sound Designer II version 2.8, the sample rate conversion for 22.050 and ll.025 are very flat and you should NOT use these post compensation setups.

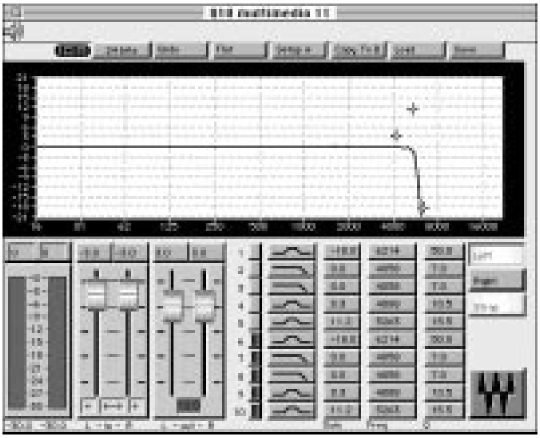
Use of setups

Not for use with Sound Designer II 2.8 and greater. Only for SDII v2.7 and below.

These setups are intended to compensate for this droop by post-processing the sample-rate converted files at the final sampling rate.



Q10 multimedia22



Q10 multimedia 11

The '22' setup file is simply a brickwall low-pass filter cutting off between 10 and 11 kHz. It is essentially flat up to 10 kHz, but falls away very rapidly above that frequency. The '11' setup is similar except that it is essentially flat up to 5 kHz and falls away rapidly at and above 5.5 kHz.

These setups have two applications:

(i) to monitor the sound of files at 44.1 or 48 kHz as they will sound when passed through the bandwidth of a 22 or 11 kHz multimedia sampling rate, and

(ii) to limit the bandwidth of a 44.1 or 48 kHz sound prior to subsequent processing for applications where the file will ultimately be converted to a 22 or 11 kHz sampling rate.

This is a desirable thing to do for two reasons:

(1) at all subsequent stages of processing, the sound will be heard -with the appropriate multimedia bandwidth, so that better production decisions can be taken.

(2) the bandlimiting will generally improve the sound of any subsequent sampling rate conversion used by removing high 'aliasing' frequencies that can degrade the sound.

As loaded, the brickwall filtering is provided by bandgroup Q10 bands 6 to 10. However, by switching an additional copy of the brickwall filter in bandgroup Q10 bands 1 to 5 in, the cut-off rate can be made extremely steep at the expense of only a small-degree of non-flatness of the in-band frequency response. In fact this cut-off is so steep that sampling rate conversion from 44.1 kHz to 22.05 kHz can be achieved after this filtering simply be discarding every second sample of the output signal file.

These setups provide:

(i) A brickwall filtering at 11 kHz (for the 22 kHz sampling rates) or 5.5 kHz (for the 11 kHz sampling rates) to minimize aliasing problems in sampling rate conversion.

(ii) A high frequency boost to compensate for the high-frequency roll-off in the Sound Designer II sampling rate conversion.

(iii) A gain reduction to prevent the significant high frequency boost from overloading and causing distortion.

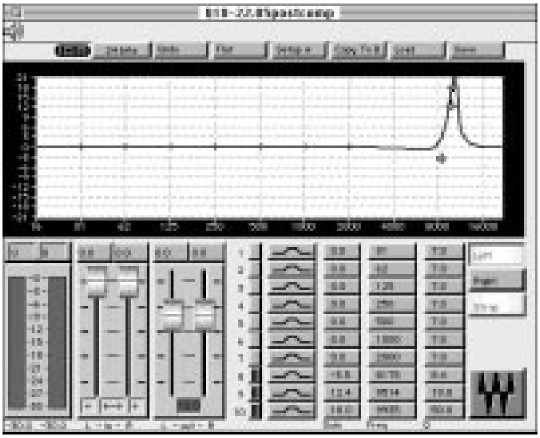
You may monitor your 22 or 11 kHz kHz files more accurately by saving copies as a Macintosh SND Resource using the Save a Copy... command, and then playing these files back using your Macintosh's audio playback via an external high-quality monitoring system.

If using Sound Manager 3.0 or above, you can play the soundfile directly through the Mac system without any filtering from SDII. From the Setup menu, select Sound Playback. Check the Sound Manager box. Now if you click on the speaker icon in SDII, the selected region will play through the Sound Manager (and out the Macintosh speaker or sound output jacks).

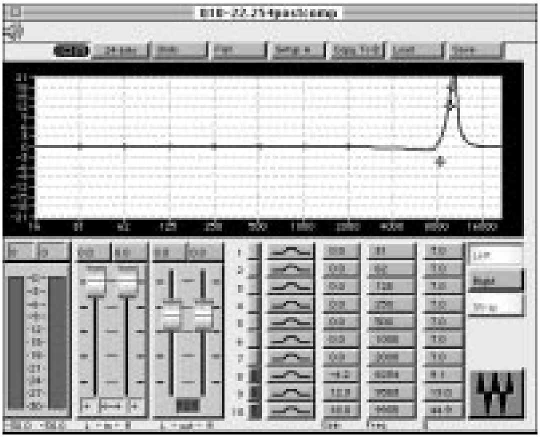
It is hoped that a simple software processing tool to accomplish this will be available in the future to allow the Q10 to be used for processing 44.1 kHz soundfiles and converting them to multimedia files at 22.05 kHz.

SDII-SR Compensation

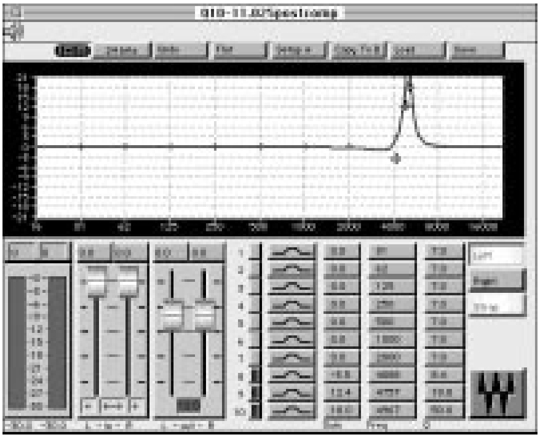
Q10 22.05postcomp
Q10 22.254postcomp
Q10 11.025postcomp



22.05postcomp



22.254postcomp



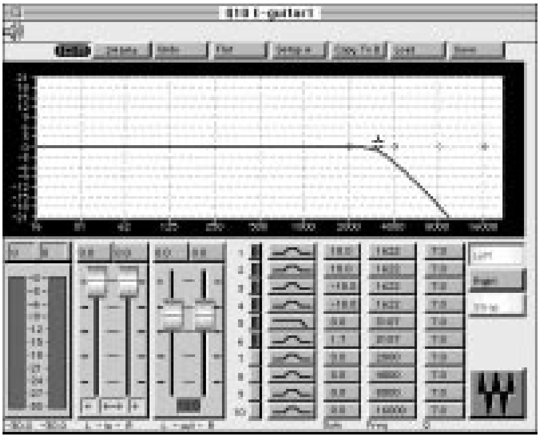
11.025postcomp

These setup files are intended to compensate for the poor frequency response of the sampling rate conversion algorithm provided with Sound Designer II **version 2.7 and below** when subsequently converting to the multimedia sampling rates of 22.05 kHz (PC) or 22.254 kHz (MAC) or 11.025 kHz from files sampled at 44.1 kHz. They typically increase the audio bandwidth at the final sampling rate by 25%, and with a generally flatter response up to cut-off.

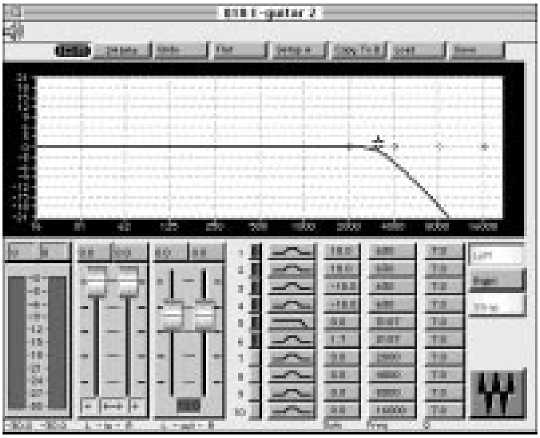
Measurement in the digital domain shows that the Sound Designer II sampling rate conversion in **version 2.7 and below** from 44.1 kHz to 22.05 kHz causes a frequency response which is about 1 dB down at 7 kHz, about 3.5 dB down at 8 kHz, about 11 dB down at 9 kHz and around 30 dB down at 10 kHz. This drooping frequency response will cause a dull sound in multimedia applications. These frequency responses occur an octave lower when a further sampling rate conversion is made from 22.05 to 11.025 kHz sampling rates using the Sound Designer II sampling rate converter.

Chapter 7 - Distortion

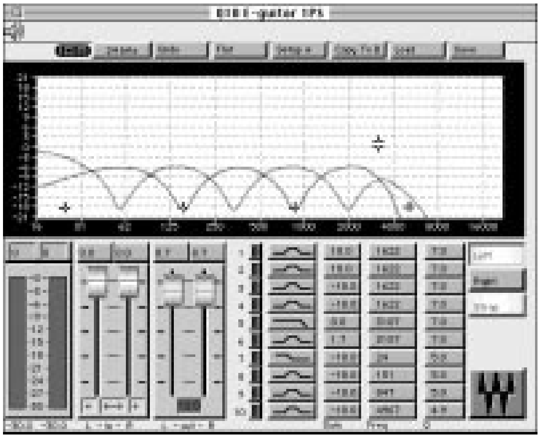
The following setups produce intentional distortion effects which may be useful in certain creative applications, both in music production and in the creation of sound effects. They are generally more effective if the input sound file is first of all passed through a dynamic compression process.



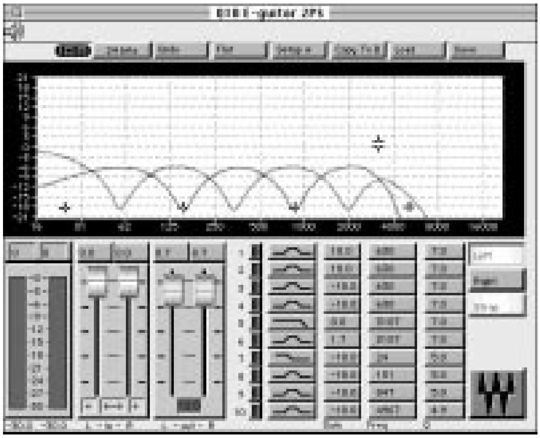
Q10 E-guitar1



Q10 E-guitar2



Q10 E-guitar1PS



Q10 E-guitar2PS

These effects produce distortion of a kind that works particularly well with electric guitars, although there may also be creative applications when used with keyboards, bass lines, or even vocals! As loaded, the first E-guitar effect produces a distortion effect that retains a lot of the clarity of the original sound, whereas the second E-guitar effect is much more dirty and 'heavy metal/grunge'.

The E-guitar distortion effect is highly adjustable and tunable. The actual distortion itself is produced by bandgroup Q10 bands 1 to 4, and the high-frequency content of the distortion is tamed by a low-pass filter bandgroup Q10 bands 5 & 6.

The effects of particular adjustments are as follows:

The distortion quality is adjusted by selecting and dragging the frequency value windows of Q10 bands 1 to 4. This selects the frequencies in the original sound that are subject to distortion, and may be tuned to get the most desirable effect from a particular instrument or sound. With bass guitar, for example, generally a frequency in the low hundreds of Hz should be used, whereas electric guitar works best with from a few hundred to two or three thousand Hz, depending on the desired effect.

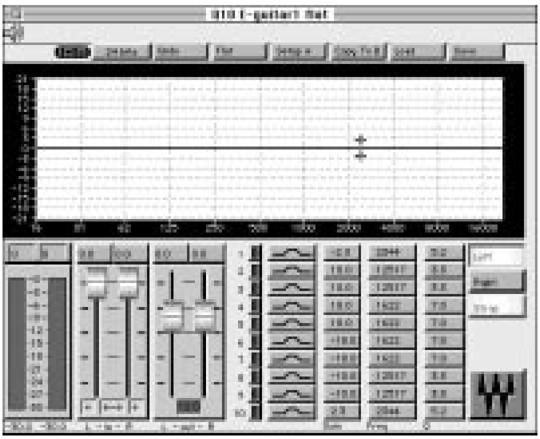
The filter may be separately tuned in frequency by selecting and dragging the frequency value windows of bandgroup Q10 bands 5 and 6, and adjusted for less or more 'toppy' distortion.

The distortion may be switched in and out without affecting the filtering by selecting and clicking the In/Out buttons of Q10 bands 1 to 4.

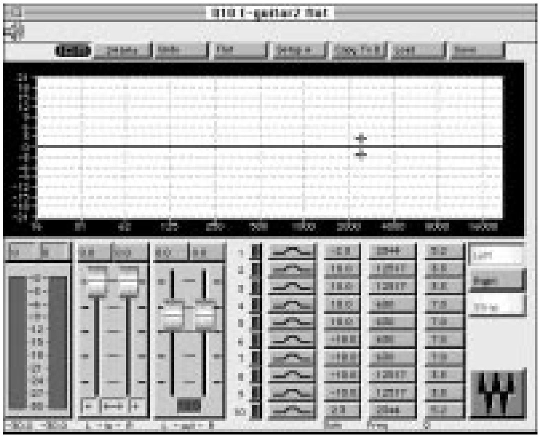
Q10 bands 7 to 10 are available for any additional EQ of the user's choice.

To reduce the amount of distortion without changing its character, reduce the input gains, and increase the output gains to obtain the required level.

The PS versions of the setups are identical except that a pseudostereo effect is added in bandgroup Q10 bands 7 to 10 to spread an input mono sound in the stereo stage. As with all pseudostereo effects, this will process a sound file with stereo output only if the input file is a stereo file with identical 'double-mono' channels.



Q10 E-guitar1flat



Q10 E-guitar2flat

These effects are guitar distortion effects similar to the previous ones, except that the low-pass filtering is applied only to the distortion, leaving the original undistorted part of the guitar sound with a flat frequency response.

This effect uses all 10 Q10 bands, and the distortion-producing bandgroup is bands 4 to 7.

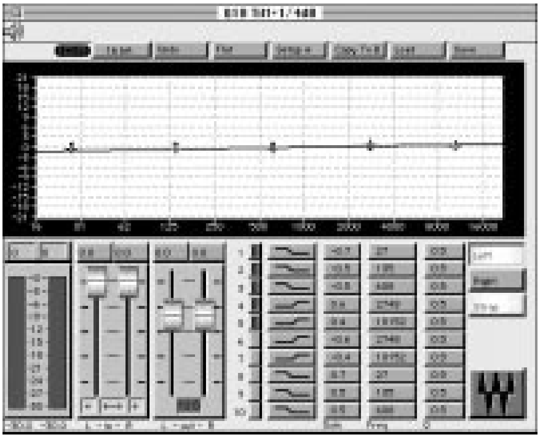
The distortion quality is adjusted by selecting and dragging the frequency value windows of Q10 bands 4 to 7. This selects the frequencies in the original sound that are subject to distortion, and may be tuned to get the most desirable effect from a particular instrument or sound. With bass guitar, for example, generally a frequency in the low hundreds should be used, whereas electric guitar works best with from a few hundred to two or three thousand Hz, depending on the desired effect.

To reduce the amount of distortion without changing its character, reduce the input gains, and increase the output gains to obtain the required level.

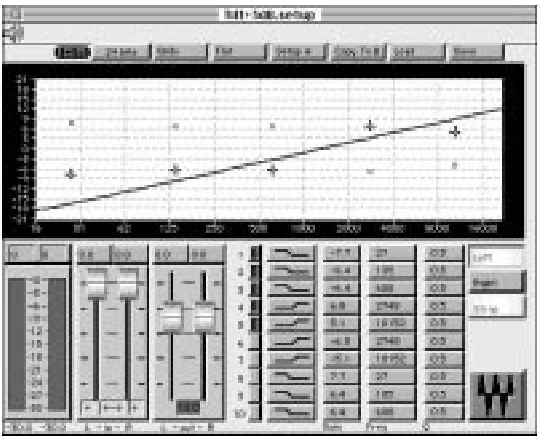
Chapter 8 - Tilt and Semi-Tilt

These setups, designed for 44.1 kHz sampling rate files, equalize sound files with a constant number of dB per octave boost or cut between 16 Hz and 20 kHz.

Bandgroup 1-5 gives a boost of the treble and cut in the bass. Bandgroup 6-10 gives a boost in the bass and cut in the treble.



Q10 Tilt+1/4dB



Q10 Tilt+3dB

The Q10 'Tilt' setup files give respective treble boosts of the indicated dB per octave if bandgroup 1-5 is switched in and treble cut the indicated dB per octave if bandgroup 6-10 is switched in.

The tilt setups are suitable for whenever an overall bass boost/treble cut or treble boost/bass cut is required without any particular frequency band being selected for special treatment. The overall effect is more “tonally neutral” than conventional treble/bass EQ, while giving an overall tonal tilt.

The **Q10 Tilt+3dB** EQ bandgroup 1-5 can be used to convert pink noise to white noise, or to convert white noise to blue noise and **Q10 Tilt+6dB** EQ bandgroup 1-5 can be used to convert pink noise to blue noise. Conversely, the **Q10 Tilt+3dB** EQ bandgroup 6-10 can be used to convert white noise to pink noise or pink noise to red noise, and the **Q10 Tilt+6dB** EQ bandgroup 6-10 can be used to convert white noise to red noise.

Bandgroup 1-5 of **Q10 Tilt+6dB** acts as an approximate differentiator within the audio band, and bandgroup 6-10 of **Q10 Tilt+6dB** acts as an approximate integrator within the audio band.

One can copy and paste two bandgroups from tilt setups to obtain other degrees of tilt. For example if one loads **Q10 Tilt+6dB**, and then copies bandgroup 1-5 of **Q10 Tilt+5dB** into bands 6-10 of **Q10 Tilt+6dB**, the overall effect of having bands 1-10 engaged will be to have a treble boost of 6+5 = 11 dB per octave. Obviously, especially with extreme degrees of boost, it may be advisable to reduce input gain until no clipping is audible, as the massive treble boosts involved will be likely otherwise to cause clipping.

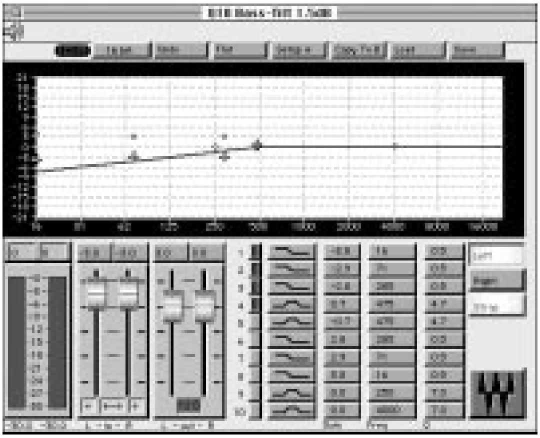
In practice, these tilt setups also work well at a 48 kHz sampling rate, but not at lower sampling rates like 22 or 32 kHz.

Sub-classes of setups:

Bass-Tilt (tilt below a frequency 250 to 1 kHz),
Treble-Tilt (tilt above 2 kHz)
and **High-Tilt** (tilt above 4 kHz)

The Bass-Tilt setups, designed for all sampling rates, equalize sound files with a constant number of dB per octave boost or cut below a "hinge frequency" pre-set at 500 Hz, although the hinge frequency may be adjusted up to about 1 kHz.

Bandgroup 1-5 gives a cut in the bass.
Bandgroup 6-10 gives a boost in the bass.



Q10 Bass-Tilt 1.5dB

These setups, designed for all sampling rates, equalize sound files with a constant number of dB per octave boost or cut below a "hinge frequency" pre-set at 500 Hz, although the hinge frequency may be adjusted up to about 1 kHz. Bandgroup 1-5 gives a cut in the bass. Bandgroup 6-10 gives a boost in the bass.

The **Q10 Bass-Tilt** setup files give indicated bass cuts in dB per octave if bandgroup 1-5 is switched in and bass boosts in dB per octave if bandgroup 6-10 is switched in.

The hinge frequency below which the bass boost or cut occurs is loaded as 500 Hz, but this may be adjusted downward to around 250 Hz and upwards up to around a kHz or so by selecting and dragging the frequency windows of bandgroups 1-10. The hinge frequency can be set even higher than 1 kHz, but the characteristic departs from pure "tilt" at very low bass frequencies if one does so.

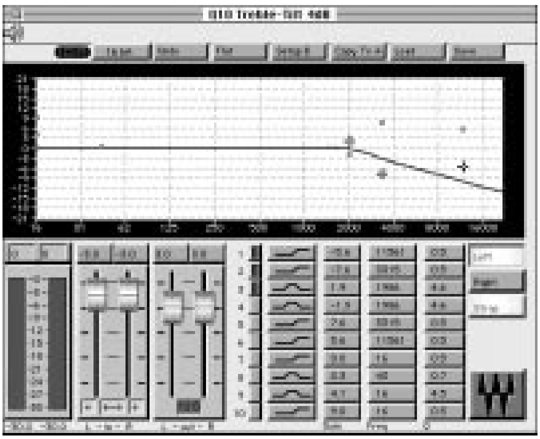
The bass tilt setups are suitable for whenever an overall bass boost or bass cut is required without any particular bass frequency band being selected for special treatment.   The overall effect is more “tonally neutral” than conventional bass EQ, while giving an overall tonal tilt.

One can copy and paste two bandgroups from bass tilt setups to obtain other degrees of bass tilt. For example if one loads Bass-Tilt 4dB into Setup A, and then loads Bass-Tilt 3.5dB into setup B and then copies bandgroup 1-5 of it into bands 6-10 of setup A, the overall effect of having bands 1-10 of setup A engaged will be to have a bass cut of 4+3.5 = 7.5 dB per octave. Obviously, especially when boosting bass, it may be advisable to reduce input gain until no clipping is audible, as the boosts involved will be likely otherwise to cause clipping.

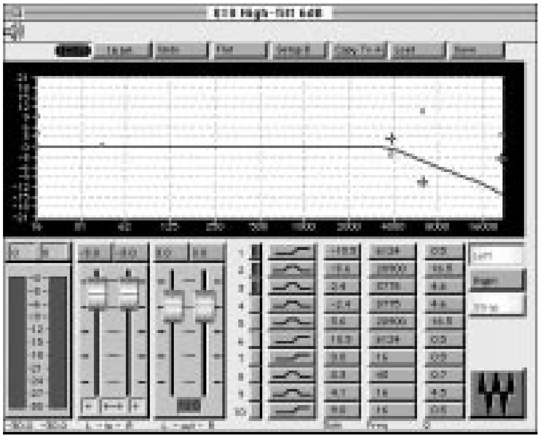
Treble and High-tilt

These Treble-Tilt and High-Tilt setups, designed for the 44.1 and 48 kHz sampling rates, equalize sound files with a constant number of dB per octave boost or cut above a “hinge frequency” pre-set at 2 kHz for the “Treble” setups and 4 kHz for the “High” setups. Bandgroup 1-3 gives a cut in the treble. Bandgroup 4-6 gives a boost in the treble.

The hinge frequency is not adjustable on these setups without altering the “tilt” shape.



Q10 Treble-Tilt 4dB



Q10 High-Tilt 6dB

The Q10 Treble-Tilt setup files give indicated treble cuts in dB per octave between 2kHz and 20 kHz if bandgroup 1-3 is switched in and treble boosts in dB per octave between 2kHz and 20 kHz if bandgroup 4-6 is switched in.

The Q10 High-Tilt setup files give indicated treble cuts in dB per octave between 4kHz and 20 kHz if bandgroup 1-3 is switched in and treble boosts in dB per octave between 4kHz and 20 kHz if bandgroup 4-6 is switched in.

The Treble- and High-tilt setups are suitable for whenever an overall treble boost or treble cut is required without any particular treble frequency band being selected for special treatment. The overall effect is more “tonally neutral” than conventional treble EQ, while giving an overall tonal tilt.

The High-Tilt setups are especially useful for subtle adjustment and enhancement or taming of high treble while having little effect on basic tonal quality, and are extremely useful for fine adjustment of lackluster or sizzly older recordings.

One can copy and paste two bandgroups from Treble- or High-tilt setups to obtain other degrees of tilt. For example if one loads **Q10 Treble-Tilt 4dB** into Setup A, and then loads **Q10 Treble-Tilt 3.5dB** into setup B and then copies bandgroup 1-3 of it into bands 4-6 of setup A, the overall effect of having bands 1-6 of setup A engaged will be to have a treble cut of 4 + 3.5 = 7.5 dB per octave above 2 kHz. Obviously, especially when boosting treble, it may be advisable to reduce input gain until no clipping is audible.

Combining Bass- and Treble- or High-tilts

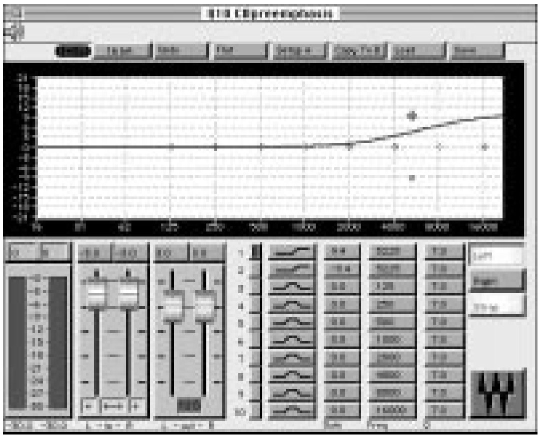
Bass-Tilts may be combined with Treble- or High-Tilts. Load the Bass-Tilt setup desired in Setup A, selecting the bandgroup of 5 bands required and adjusting its hinge frequency as required, then load the Treble-Tilt or High-Tilt setup desired in Setup B. Select the bandgroup of 3 bands required; copy/paste it into an unused slot of 3 bands in Setup A. If Bass-Tilts with hinge frequency lowered to around 250 Hz are combined with High-Tilts, this can result in a tonally neutral effect combined with quite subtle alterations of bass and treble feel.

Chapter 9 - Pre/De-Emphasis

These setups are intended to add or remove various standard pre-emphasis curves from sound files in the digital domain. Since these are actually digital IIR filters, they are not quite exact simulations of the analog filters that define the curves, but they are usually very close.

It is important when using pre-emphasis (and sometimes de-emphasis) to reduce the input gains when necessary to prevent overload and clipping of the output.

Q10 CDpreemphasis

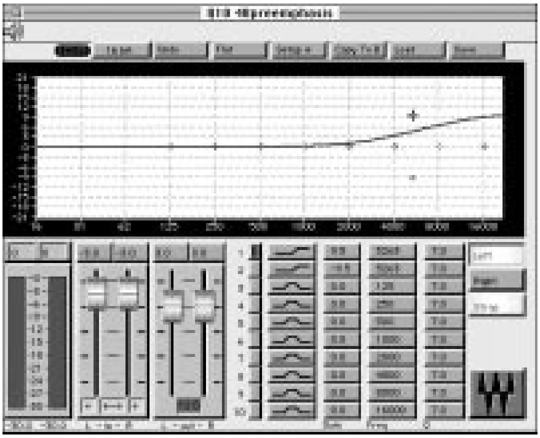


This setup approximates the standard CD and DAT pre-emphasis curve (with time constants 50 and 15 microseconds) at a sampling rate of 44.1 kHz, to within about 0.05 dB up to 16 kHz.

Band 1 alone implements the CD pre-emphasis on sound files recorded without pre-emphasis.

Band 2 alone implements the inverse de-emphasis, and may be used to remove pre-emphasis on a sound file having pre-emphasis.

Q10 48preemphasis

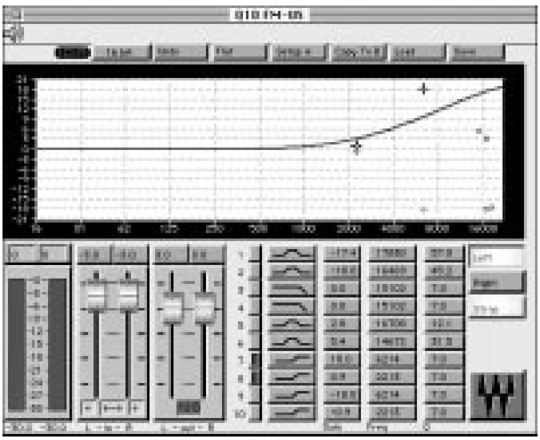


This setup approximates the standard CD and DAT pre-emphasis curve (with time constants 50 and 15 microseconds) at a sampling rate of 48 kHz, to within about 0.05 dB up to 18 kHz.

Band 1 alone implements the pre-emphasis on sound files recorded without pre-emphasis.

Band 2 alone implements the inverse de-emphasis, and may be used to remove pre-emphasis on a sound file having pre-emphasis. It is useful for removing pre-emphasis from tapes recorded on DAT machines using pre-emphasis.

Q10 FM-US
Q10 FM-US48



US FM pre-emphasis

This setup approximates the standard US FM broadcast pre-emphasis curve (with time constant 75 microseconds) at a sampling rate of 44.1 kHz (for Q10 FM-US) or 48 kHz (for Q10 FM-US48), to within about 0.1 dB up to 15 kHz.

The standard US FM pre-emphasis is also used for FM broadcasting in most of the Americas and in Japan.

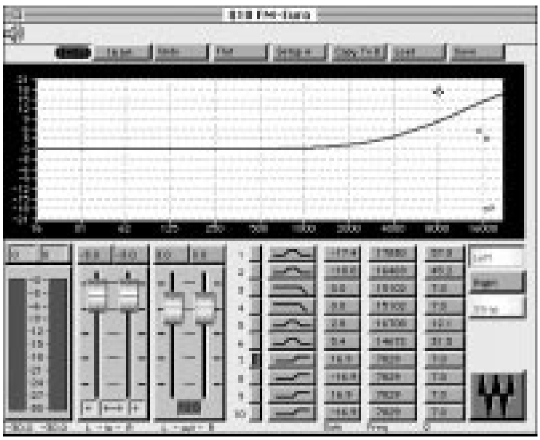
Bandgroup 7 & 8 implements the FM pre-emphasis on sound files recorded without pre-emphasis.

Bandgroup 9 & 10 alone implements the inverse de-emphasis, and may be used to remove pre-emphasis on a sound file having pre-emphasis.

Bands 1-6 implement a brickwall low-pass filter cutting off at 15 kHz - the standard bandwidth limitation used in FM broadcasting.

A typical use would be to peak-limit sound files for FM broadcasting. To do this, first process a sound file by using on of two FM pre-emphasis setups, engaging bands 1-8, with a gain reduction on the inputs to avoid clipping. Then pass the resulting sound file through the Waves L1 limiter to limit levels of the pre-emphasized filtered signal. Then pass the resulting signal through bandgroup 9 & 10 ONLY of this setup of the Q10 to remove the pre-emphasis. The resulting file will then be limited in a way suited for FM broadcast.

Q10 FM-Euro
Q10 FM-Euro48



European FM pre-emphasis

This setup approximates the standard European FM broadcast pre-emphasis curve (with time constant 50 microseconds) at a sampling rate of 44.1 kHz (for Q10 FM-Euro) or 48 kHz (for Q10 FM-Euro48), to within about 0.1 dB up to 15 kHz.

Band 7 alone implements the FM pre-emphasis on sound files recorded without pre-emphasis.

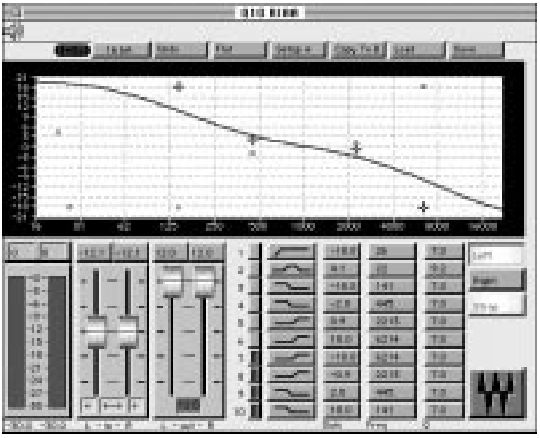
Band 8 alone implements the inverse de-emphasis, and may be used to remove pre-emphasis on a sound file having pre-emphasis.

Bands 1-6 implement a brickwall low-pass filter cutting off at 15 kHz - the standard bandwidth limitation used in FM broadcasting.

A typical use would be to limit sound files for FM broadcasting. To do this, first process a sound file by using FM pre-emphasis engaging bands 1-7 of this setup, with a gain reduction on the inputs to avoid clipping. Then pass the resulting sound file through the Waves L1 limiter to limit levels of the pre-emphasized filtered signal. Then pass the resulting signal through band 8 ONLY of this setup of the Q10 to remove the pre-emphasis. The resulting file will then be limited in a way suited for FM broadcast.

Q10 RIAA

Q10 RIAA48



RIAA de-emphasis for phonograph records

This setup approximates the standard RIAA phongraph LP de-emphasis and pre-emphasis curves (with time constants 3180, 318 and 75 microseconds) at a sampling rate of 44.1 kHz (for Q10 RIAA) or 48 kHz (for Q10 RIAA48), to within better than 0.1 dB up to 15 kHz.

Its primary use is to equalize to flat soundfiles obtained by playing LP or 45 rpm phonograph record played on a velocity-sensitive cartridge into a pre-amp not having the RIAA equalization. It allows use of soundfiles from phonograph records without the need for an equalized pre-amplifier input. For the most accurate results, one should take care that the pre-amplifier input has an impedance matched to the load impedance recommended by the manufacturer of the phonograph cartridge used.

Bandgroup 1-2 is a rumble high-pass filter, set for -3dB point at about 20 Hz. This may be used to remove infrasonic rumble from soundfiles taken from phonograph records. Its operating frequency may be adjusted by selecting and dragging together the two frequency windows of this bandgroup. Use of this filter minimizes the risk of very high levels of infrasonic rumble once RIAA de-emphasis is used.

Bandgroup 7-10 alone implements the RIAA de-emphasis on sound files recorded with pre-emphasis, such as the output of a velocity phonograph cartridge feeding an unequalized pre-amplifier input. It may be used to equalize the sound from phonograph records to flat response. When this de-emphasis is used, take care to reduce the input gains, and also if necessary the output gains, to prevent output clipping

Bandgroup 3-6 alone implements the inverse RIAA pre-emphasis, and may be used to give RIAA pre-emphasis to a sound file having no pre-emphasis. There may not be much practical us for this, but it could be useful for the study of the effects of RIAA pre-emphasis.

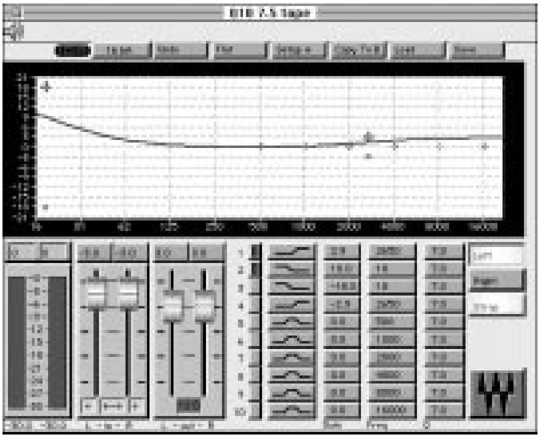
One application is when older records such as 78 rpm records or early LPs have been played via a standard RIAA de-emphasis phongraph cartridge stage. You can engage the pre-emphasis Q10 bands 3-6 to remove the unwanted RIAA de-emphasis before adding a new de-emphasis for an older record de-emphasis characteristic.

Hint: For most older 78 RPM records pre-dating the late 1940s, a good approximation to older playback characteristics with constant velocity above a hinge frequency and constant amplitude at lower frequencies can be obtained as follows: Play the record over a conventional RIAA pre-amplifier stage and convert as usual to the digital domain. Engage Q10 bands 3-6 to remove the unwanted RIAA de-emphasis. In addition, engage Q10 bands 9 &10 to reintroduce constant amplitude characteristics at lower frequencies, BUT adjust the "hinge frequency" of this characteristic up and down by ear for best results by selecting the two frequency value windows of bandgroup 9 &10, and dragging these up or down. Q10 bandgroup 1 & 2 may be switched in or out to reduce low frequency rumble, and the frequency of the rumble filter may be adjusted as before by selecting and dragging the two frequency value windows of Q10 bandgroup 1 & 2. Q10 bandgroup 7 & 8 should remain switched out in this 78 rpm record re-equalization, or they may be used for additional conventional parametric equalization to the user's taste for additional tonal correction or for filtering "scratch noise".

Beyond what is said here, re-equalization of older 78 rpm records is a specialist skill, and there is no substitute for extensive experience for best results. The skills vary from finding the correct playback speed (older records were often recorded at other speeds), to the use of special stylus sizes and shapes (never use an LP stylus!) matched to the grooves of records from different times and places. With correct playback methods and skilled adjustment of equalization characteristics as above, one is often surprise to find just how very good old recordings can be–far better than is often realized when played back incorrectly. The natural sound quality of many old recordings can still put to shame many modern studio productions.

In order to avoid clipping of the Q10 equalization in this 78 rpm application, it may be advisable to reduce the input gain.

Q10 7.5 tape
Q10 15 tape



Conversion between IEC and NAB analog tape playback

Analog reel-to-reel tape at 7.5 or 15 inches per second speeds in the USA was largely recorded according to NAB playback equalization characteristics with time constants 3180 and 50 microseconds at both speeds, whereas in Europe, tapes were largely recorded for playback with IEC playback equalization characteristics with time constant 70 microseconds at 7.5 inches per second and 35 microseconds at 15 inches per second.

While many tape recorders offer playback with either NAB or IEC characteristics, other machines offer only NAB or only IEC playback, and will play tapes recorded with the other characteristic incorrectly. These setups provide correction of the equalization error caused by playing back tapes on the wrong equalization setting.

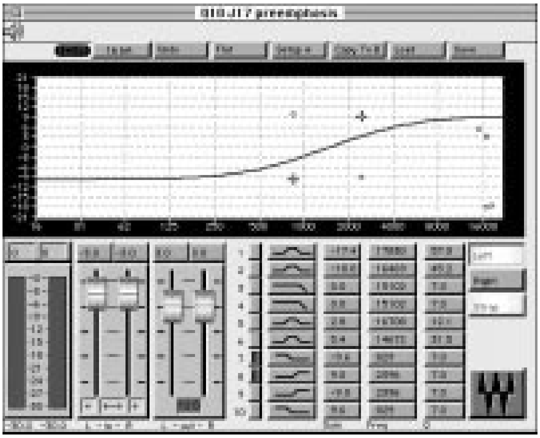
To correct European IEC tapes played back on NAB tape machines, select the setup for the tape speed (7.5 or 15). Select bands 1 and 2 only.

To correct US NAB tapes played back on IEC tape machines, select the setup for the tape speed (7.5 or 15). Select bands 3 and 4 only.

WARNING: These correction equalizations should only be used if no noise reduction is employed on the tape. Noise reduction will work correctly only if tapes are played on machines using the correct playback equalization.

These correction setups work at both 44.1 and 48 kHz sampling rates.

Q10 J17 pre-emphasis



There are two internationally standardized pre-emphasis and de-emphasis characteristics for digital audio. The best known is the 50/15 microsecond characteristic sometimes used with CD and 48 kHz DAT tapes. However, there is another characteristic known by the technical name CCITT J17, widely used in broadcasting applications especially at a sampling rate of 32 kHz. J17 pre-emphasis incorporates a deep bass cut as well as a treble boost.

J17 has two time constants, 333.33 and 38.49 microseconds.

To add J17 pre-emphasis to a sound file without pre-emphasis, engage Q10 bandgroup 7 & 8 only.

To remove J17 pre-emphasis from a J17 pre-emphasized sound files, engage Q10 bandgroup 9 & 10 only. This implements J17 de-emphasis.

The J17 equalization will work well at sampling rates between 22 and 48 kHz, although its accuracy is optimized for the 44.1 kHz sampling rate.

This setup also includes a "brickwall" steep 15 kHz cut-off filter as bandgroup 1-6 to reduce the bandwidth to that used in broadcast NICAM applications. This filter is specifically optimized for the 44.1 kHz sampling rate, and SHOULD NOT be used at significantly lower sampling rates such as 32 or 22.05 kHz. The filter also still works reasonably well at the 48 kHz sampling rate.

On some sound files, clipping may occur, and if so, reduce the input gain. If the output is not fully modulated, some increase in gain may be desired–increase gain using the output faders to minimize risk of internal clipping in the Q10. The gain settings given are found to work well on many sound files, but may not always be best.

J17 is widely used at the 32 kHz sampling rate in broadcasting, and is the standard pre-emphasis used with the NICAM system of digital TV sound broadcasting in many parts of Europe, including the UK. Versions of NICAM are also widely used for digital distribution of broadcast sound signals both for TV and FM broadcasting, and has been used since 1972.

This setup can be used to convert conventional sound files to J17 pre-emphasis at a sampling rate of 32 kHz. Such sound files will be compatible with NICAM applications.

A typical use of this setup would be to limit sound files for broadcasting via NICAM or where NICAM links are used. To do this, first process a sound file by using NICAM pre-emphasis engaging bandgroup 7 & 8 of this setup, with a gain reduction on the inputs to avoid clipping if necessary. If the soundfile is at a sampling rate of 44.1 or 48 kHz, you may also wish to engage bandgroup 1-6 to remove frequencies above 15 kHz that are not wanted in the broadcast application. Then pass the resulting sound file through the Waves L1 limiter to limit levels of the pre-emphasized filtered signal. Then pass the resulting signal through bandgroup 9 & 10 of this setup of the Q10 to remove the pre-emphasis. The resulting file will then be limited in a way suited for NICAM broadcast.

Comments on other uses of J17 pre-emphasis

J17 is in theory internationally standardized, alongside the standard pre-emphasis used with CD and DAT, as a pre-emphasis for use with all digital systems, although it is rarely implemented except at a sampling rate of 32 kHz. Louis Fielder of Dolby Laboratories, noted authority on dynamic range issues, has argued that use of J17 pre-emphasis, unlike conventional CD pre-emphasis, has the effect of increasing the real dynamic range that can be handled by digital channels on real musical signals. The availability of the J17 emphasis (accurate to within about 0.1 dB up to 20 kHz at the 44.1 kHz sampling rate) will allow users to evaluate for themselves its performance, especially when the pre-emphasized files are quantized to low-bit wordlengths.

It is suggested that J17 may be ideal for handling multimedia files at sampling rates of 22.05 kHz, if played back via the corresponding de-emphasis. Indeed, J17 would make a good de-emphasis characteristic for multimedia files for lower noise and distortion. Fielder has also argued on the basis of extensive measurements of peak signal characteristics that J17 is also ideal for maximizing signal-to-noise ratio for professional applications at 44.1 and 48 kHz sampling rates, and would have made a far better pre-emphasis characteristic for CD than the one actually used.

The deep bass cut means that on many typical sound files having predominant energy below 1 kHz, the amplitude of the bass is greatly cut, and this allows the treble to be boosted considerably without the overall amplitude increasing. The inverse de-emphasis boosts bass and cuts treble, including any audible noise (including quantization noise) in the pre-emphasized signal channel. J17 is much better matched to many signals than other pre-emphasis characteristics.

Reference

L. Fielder "Dynamic Range Issues in the Modern Digital Audio Environment", Proceedings of the AES UK Conference "Managing the Bit Budget", (1994 May) pp. 3-19

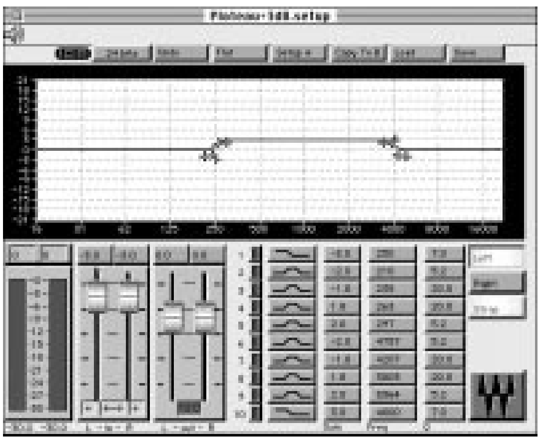
Chapter 10 - Plateau

These setups are intended to provide detailed equalization of specific frequency bands without affecting other neighboring bands. They allow achievement of a “plateau” frequency response that is raised or lowered by a given number of dB within a user selected band relative to the rest of the audio spectrum. This plateau has steep sides (the transition between the gains takes about one fifth of an octave).

Plateau equalization is found particularly useful in bringing out or subduing particular instruments within a mix without too much effect on the tonal quality of the rest of the mix, so is particularly useful for subtle corrective rebalancing work. By allowing precise reduction or increase of particular frequency bands, it also allows one component of a mix to be reduced in spectral level in a critical frequency range to leave “space” for another component of a mix to be more audible, again without too much effect on overall sound of the mix.

Q10 Plateau+9dB

Q10 Plateau-3dB
Q10 Plateau+7.5dB
Q10 Plateau-4.5dB
Q10 Plateau+6dB
Q10 Plateau-6dB
Q10 Plateau+4.5dB
Q10 Plateau-7.5dB
Q10 Plateau+3dB
Q10 Plateau-9dB
Q10 Plateau+1.5dB
Q10 Plateau-18dB
Q10 Plateau-1.5dB



Q10 Plateau+3DB

The plateau equalizations are categorized by the relative level of the plateau relative to the rest of the audio band from +9 dB to -9 dB in 1.5 dB steps, plus a -18 dB plateau. Bandgroup Q10 bands 1 to 5 control the lower frequency step, and the frequency may be altered by select-and-dragging the frequency value windows of this bandgroup. The lower transition frequency is shown in the frequency value window of Q10 band 1. Bandgroup Q10 bands 6 to 10 control the higher frequency step, and the frequency may be altered by select-and-dragging the frequency value windows of this bandgroup. The higher transition frequency is shown in the frequency value window of Q10 band 10.

As loaded, these setups have transition frequencies 250 Hz and 4 kHz, giving a 4-octave wide plateau, but by altering one or both of these frequencies, the plateau may be varied in width from a fraction of an octave to many octaves. By this means, the plateau EQ can be applied to any desired frequency range.

Bandgroup Q10 bands 6 to 10 may be switched out if one just wishes for a single step transition up or down to the plateau.

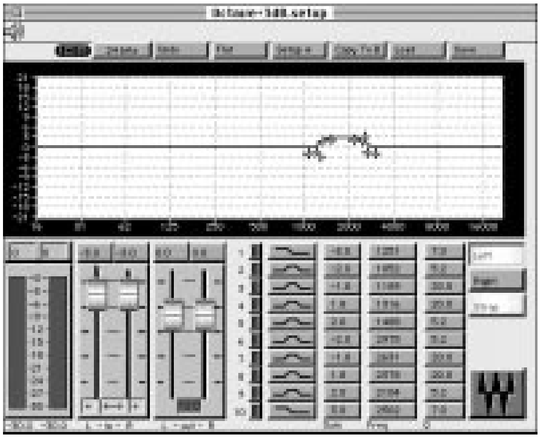
One can even paste bands 1 to 5 from one Plateau setup with bands 6 to 10 from another Plateau setup in order to have different transition step sizes before and after the plateau. For example if band 1-5 are taken from the **Q10 Plateau+6dB** setup and bands 6-10 from the **Q10 Plateau+3dB** setup, then the first step will be from 0 to + 6 dB, and the second from +6 dB to +3 dB. Input and output gains should be adjusted to avoid overload when doing this.

By this means, the plateau setups allow arbitrary adjustment of two step transition frequencies and the step sizes (in 1.5 dB steps from -9 dB to +9 dB). This effectively allows equalization of the level of 3 frequency bands virtually independently of one another.

Octave setups

Q10 Octave+9dB
Q10 Octave-3dB
Q10 Octave+7.5dB
Q10 Octave-4.5dB
Q10 Octave+6dB
Q10 Octave-6dB
Q10 Octave+4.5dB
Q10 Octave-7.5dB
Q10 Octave+3dB
Q10 Octave-9dB
Q10 Octave+1.5dB
Q10 Octave-18dB
Q10 Octave-1.5dB

These are essentially identical to the Plateau EQ's except that, as loaded, these setups have transition frequencies 1251 Hz and 2502 kHz, giving an octave wide plateau.

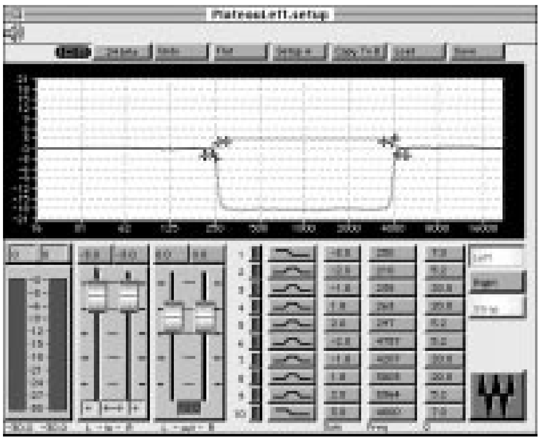


Q10 Octave +3dB

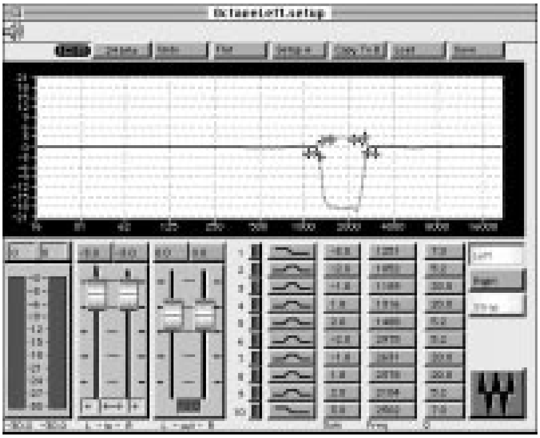
This is a typical frequency range for picking out particular instruments, this particular octave being appropriate for electric guitars.

Plateau L/R types

Q10 PlateauLeft
Q10 PlateauRight
Q10 OctaveLeft
Q10 OctaveRight

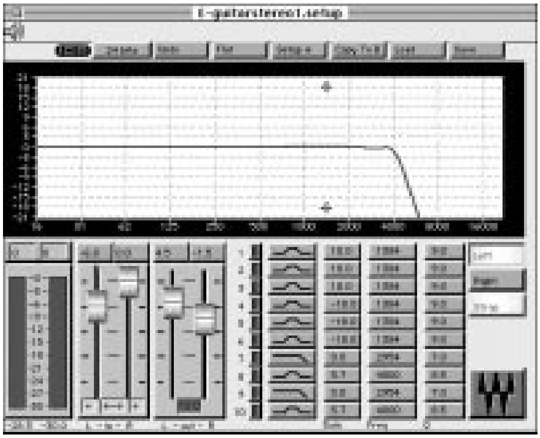


Q10 PlateauLeft
Q10 PlateauRight (not pictured)

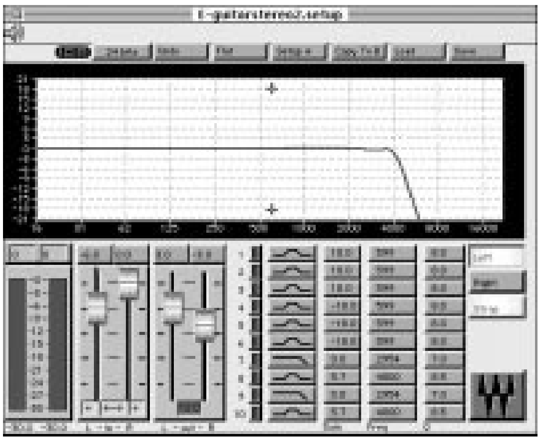


Q10 OctaveLeft
Q10 OctaveRight (not pictured)

Intended to be used with mono or double-mono signals, these four setups provide different plateau EQ on the left and the right channels. For the Left setups, the level of the left channel is boosted 3 dB and the right channel cut 18 dB in the plateau frequency region, The channels are the other way round for the Right setups. These setups in effect pan to the left (for Left setups) or right (for Right setups) over the plateau frequency region, and to the center at



Q10 E-guitarstereo1



Q10 E-guitarstereo2

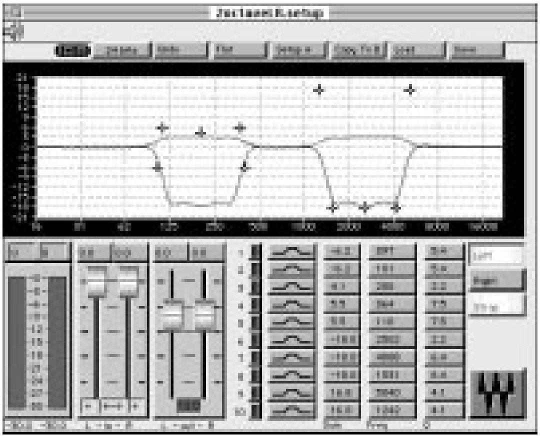
These are stereo distortion setups suitable for compressed electric guitar, and possibly bass guitar.

There are two parameters that can be tuned to alter the effect. The first is to select and drag (with linked stereo channels), the frequencies of bandgroup 1-6. This changes the quality of distortion. Higher frequencies, as in Q10 E-guitarstereo1, give a more “sizzly” effect, lower frequencies, as in Q10 E-guitarstereo2, a more “grungy” effect.

The second adjustment is to select and drag the frequencies of bandgroup 7-10. This adjusts the cut-off frequency in the treble of the guitar distortion effect.

One can also switch out bandgroup 9-10 in order to have less filtering of treble.

Q10 1.5octaveLR
Q10 2octaveLR
Q10 2.7octaveLR



Q10 2octaveLR

These pseudostereo setups are designed to pan sharply defined frequency bands of sound to the left and to the right, with other sounds in the middle. They are designed for use with mono signals presented as double mono sound files. They differ from previous setups such as **Q10 PStunable**-types. These provide very sharply defined and relatively wide frequency bands in which sounds are panned to one side. The options are similar except that they differ in the width of the panned frequency bands, respectively 1.5, 2 and 2.7 octaves.

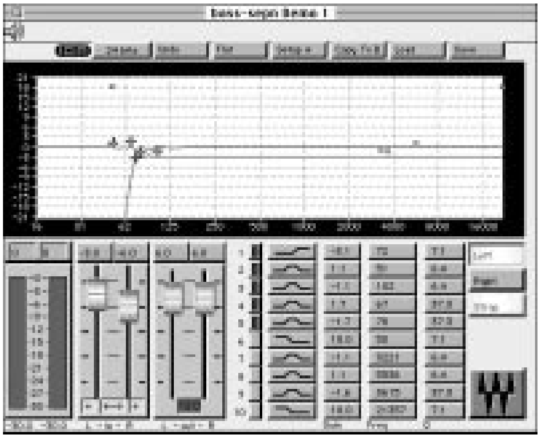
The bandgroup Q10 bands 1 to 5 controls the frequency band panned to the left. The bandgroup Q10 bands 6 to 10 controls the frequency band panned to the right. The panning of either bandgroup may be switched In/Out by selecting all 5 In/Out buttons and clicking. The frequency of operation of either bandgroup may be changed by select and dragging all 5 frequency value windows. The two bands may be in either order, i.e. bandgroup 1-5 may be at a lower frequency (as loaded) or at a higher frequency than bandgroup 6-10, but the two bands should not overlap too much–there is too much overlap if the left and right curve of the graphic display cross over much below 0 dB.

One may also have the two bands of different widths by pasting the bandgroup 6-10 of one of these setups to replace bandgroup 6-10 of another.

In use, this pseudostereo effect is particularly useful if the two bands are tuned so that the frequency ranges of particular instruments are panned to the two sides with the rest of sounds from the middle. This can often be used to synthesize a quite convincing stereo effect from mono source materials. Applications include for example historical mono material, mono PA feed signals from live recordings using PA and live ambient microphone signals, and mono ambience tracks in documentary and ENG recordings where ambiences may only have one track available. Mono compatibility should always be checked when this is important, since this effect is not always mono compatible.

Chapter 13 - Bass separation

Here are three demonstration files illustrating the effect of separation in the extreme bass.



Q10 bass-sepn Demo 1
Q10 bass-sepn Demo 2 (not pictured)
Q10 bass-sepn Demo 3 (not pictured)

It is commonly believed that the ears do not hear a stereo directional effect in the extreme bass, so that extreme bass can be fed to super-woofers placed almost anywhere in a room without degraded stereo effect. These demonstration files allow you to judge the truth or otherwise of this hypothesis for yourself!

All three files accept as input a dual-mono stereo file, and give as output a file in which the very low bass is panned to one stereo position and all other frequencies to another.

Demos 1 and 2 pan the bass to the left and other frequencies to the center. By switching bypass in and out, you can hear the effect of bass frequencies being removed from the center and panned to the left. The two demo setup files differ in the crossover frequencies used, which are respectively 75 Hz and 50 Hz approximately.

Demo 3 has crossover frequency of 80 Hz approximately, and places extreme bass at the center and other frequencies in the left.

All 3 demos give effectively constant power gain into the room at all frequencies.

The demos 1 & 2 use bandgroup 1-5 and demo 3 uses bandgroup 6-10, and the cross-over frequency may be adjusted by selecting the 5 frequencies in both channels in this bandgroup and dragging them up or down.