

BaxterEQ

MANUAL

revision 1.0

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1 Introduction

1.1. License

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1.2. Installation

Requirements:

- Win32 compatible system with SSE2 (or higher) instruction set support
- Tested and known to work in many VST compatible hosts

Put the DLL file contained in this archive in the VST plug-in folder of your host.

1.3. Overarching topics

Warning: Lower your listening volume while operating the plug-in to avoid hearing damage or damage of speakers or any other equipment.

Note: The plug-in contains a separate on/off switch (labeled 'IN') which must be powered on to run the module.

Usage tips:

- Use the 'IN' switch to toggle the plug-in on/off for A/B comparisons
- Use <ctrl> + mouse left click on a knob or switch to restore default position
- Use <shift> + mouse left click on a knob to fine adjust values
- Use this plug-in as an insert effect in any (stereo) channel of your VST host

1.4. Credits

Thanks to Patrick Barca for contributing his out-standing UI artwork again!

Special thanks to “Andrew J” as well as to “Tor / www.audioteknikk.net” for sharing ideas, insights and DSP schematics!

And many thanks to all the beta testers of course.

2 Overview

2.1. *BaxterEQ* at a glance

BaxterEQ – transparent mastering and mix buss shelving EQ

Finest tonal sweetening and finishing which always stays true to the source

- natural and accurate bass response
- authentic analog style HF curve rendering
- smoothest shelving operation

Perfectly suited for the mastering chain

- stepped controls throughout for repeatability and matched channel operation
- full dual channel layout
- full mid-side encoding support
- per channel level control for easy A/B match

Artifact free technical design

- low ripple and distortion filter implementations
- 64bit floating point internal processing
- oversampled for superior impulse response

Meticulously selected frequencies

- Baxandall shelving filters
 - LF @ 74, 84, 98, 116, 131, 166, 230 and 361 Hz
 - HF @ 1.6, 1.8, 2.1, 2.4, 3.4, 4.8, 7.1, 11 and 18 kHz
- 2-pole Butterworth filters
 - LC @ 12, 18, 24, 30, 36, 43 and 54 Hz
 - HC @ 7.5, 9, 11.1, 12.6, 16, 21, 28 and 40 kHz

In this device, an additional analog signal path emulation provides some subtle but precious stereo imaging improvements.

2.2. Reference



From left to right:

- M/S switch – turns the internal mid-side encoding on or off
- CUT – sets the frequency of the low-cut filter option
- SHELF – sets the frequency for the LF shelving filter
- Link switch – in upper position both channels LF controls are linked
- LF – sets the amount of shelving operation for the LF shelf
- HF – sets the amount of shelving operation for the HF shelf
- SHELF – sets the frequency for the HF shelving filter
- Link – in upper position both channels HF controls are linked
- CUT – sets the frequency of the high-cut filter option
- VOL – per channel output volume adjustment
- IN switch – power on/off

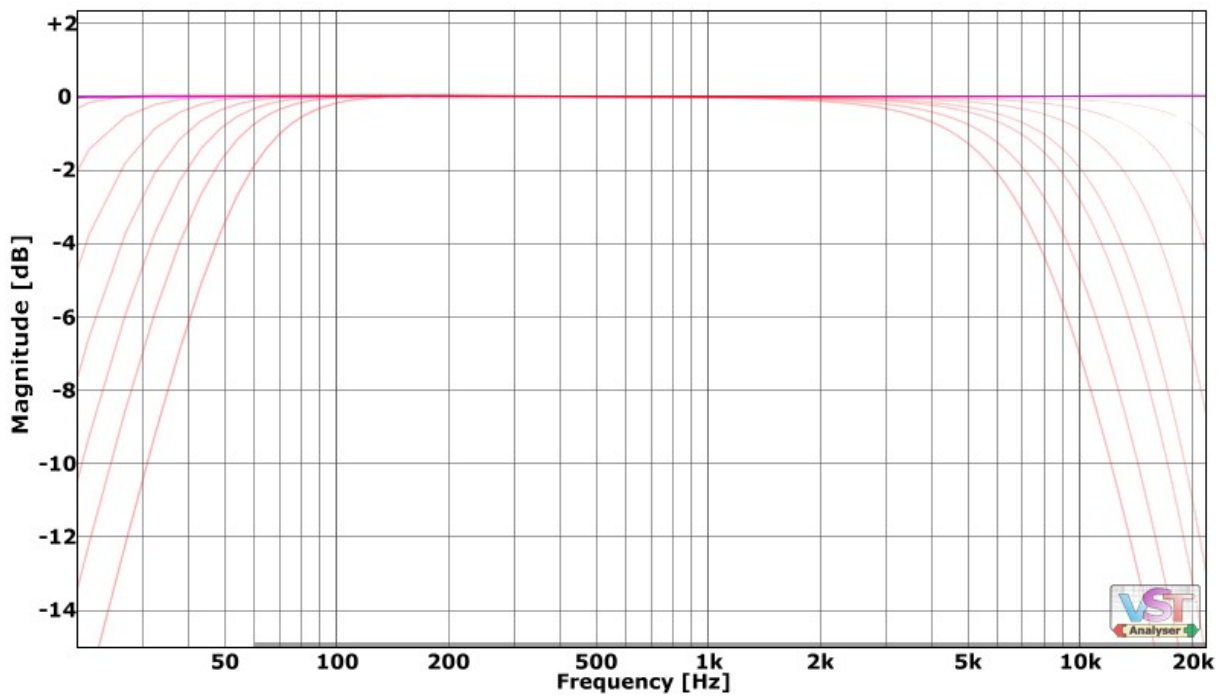
2.3. Basic workflow

For easiest workflow it's recommended to always start in L/R mode, link both channels control sets and apply overall shelving and cutting as needed. Then, unlink both control sets, switch to mid-side mode to refine EQing per channel and adjust the output volume control per channel as well.

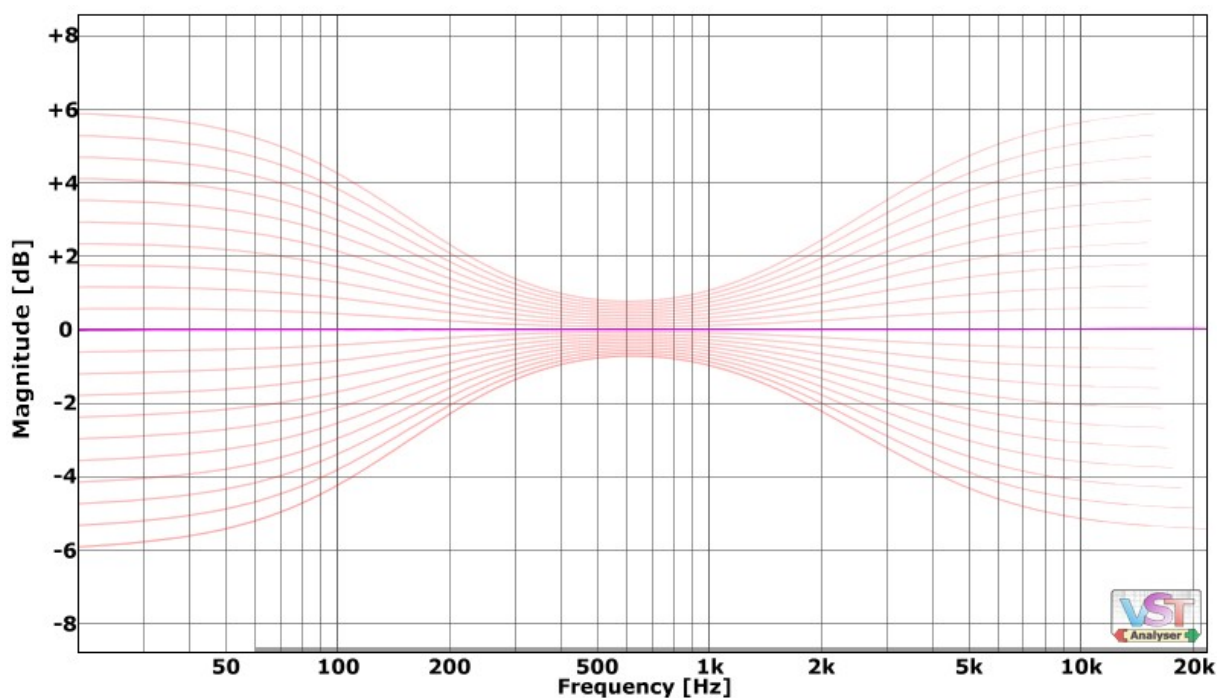
3 Addendum

3.1. Reference plots

The cut filters:



Example shelving operation (showing LF at 74 Hz and HF at 2.4 Hz):



3.2. The mid-side technique

The mid-side (M-S) stereo technique is one of the two formats of "intensity stereo," that is, stereo in which spatial localization is determined by the differences in the intensity of a sound wave as it arrives in phase at a coincident pair of microphones. Intensity stereo relies completely on the directional characteristics (polar patterns) of the microphone pair to produce this effect, since only intensity differences and not phase differences exist between the channels for any single source arriving at a coincident pair.

(Source: "M-S Stereo: A Powerful Technique for Working in Stereo" by Wesley L. Dooley and Ronald D. Streicher)

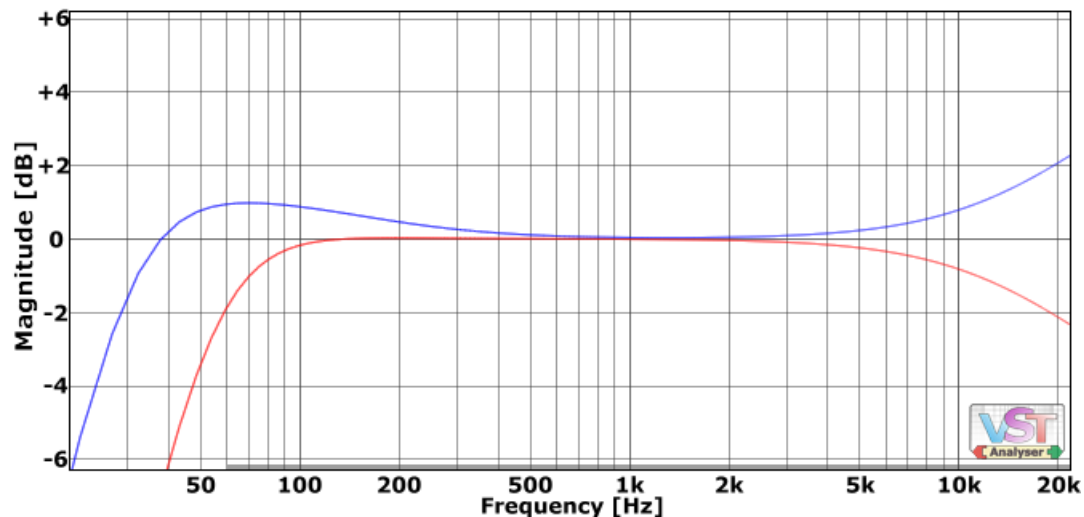
The most common situations where M/S (aka M-S aka mid-side) techniques are getting applied are:

- during the recording process when mid-side microphonie is utilized
- on the 2bus during the late mixing or mastering stage
- in audio restauration situations

I'm skipping the mid-side microphonie stuff here and just recommend "A More Realistic View of Mid/Side Stereophony" by Trevor Owen de Clercq which is available online at http://www.midside.com/pdf/nyu/masters_thesis.pdf.

While the M/S signal is obtained in a natural fashion during microphone stereophony, in most other cases stereo (L/R) encoded signals are available. In this case and to obtain the advantages of M/S a conversion is necessary.

The conversion is dead easy and is accomplished via a sum-and-difference matrix network, where typically the mid signal is the sum ($M = R + L$) and the side signal is the difference ($S = R - L$). The other way around is that simple as well and there are plenty of tools available which handle this stuff for us in the digital audio workstation.



treating two channels frequency response differently

Once a stereo source is encoded in M/S, the door is open to treat mid- and side content individually. For example, this allows for selective correction of some problems encountered on location such as out-of-phase low-frequency noise from the environment. In this case the side channel, which contains the majority of this information, can be passed through a high-pass filter to reduce such unwanted low-frequency content, and this can be done without any alteration of the mid content.

Some other typical mixing/mastering targets ideally achieved in the M/S domain are:

- assuring mono compatibility
- stereo widening and increasing depth perception
- attenuating or emphasizing the signals room information
- increasing intelligibility of voice in a mix

This is accomplished mostly by performing alterations of both channels frequency and dynamic response. The tools to be used just need to have separate controls per channel as long as M/S encoding/decoding is done externally before and after the plugin, other tools already offer internal M/S handling to make things easier: The BaxterEQ is an example for a M/S frequency shaper and Density MKII is an example for a M/S dynamics compressor.

3.3. Updates and further information

Refer to my Blog at <http://varietyofsound.wordpress.com> for some additional information and updates on this plug-in or leave a note there if any issues did occur.

Peace,
Herbert