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ALPHA COMPRESSOR


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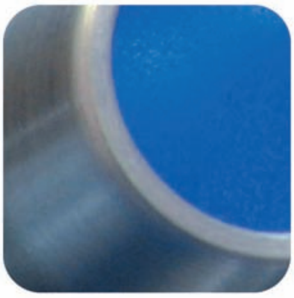



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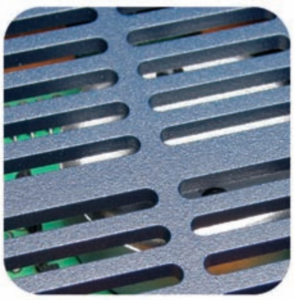


RoHS compliant 2002/95/EC

This product is manufactured according to the 2002/95/EC directive. The purpose of this directive of the European Union is the Restriction of Hazardous Substances (RoHS) in electronic equipment in order to protect health and nature.

**WARNING: High Voltage**

- Risk of electric shock.
- Do not open chassis.
- Refer service to qualified service staff only.
- Before connecting the device to the main power supply, check if the right voltage is selected.
- Replace fuse with the same type and value only.
- This device must be connected to ground.
- Do not use a damaged power cord.
- Never place containers with liquid, e.g. beverages or a vase, on the unit.
- Do not expose this device to rain or moisture.
- Do not use this device near water, e.g. swimming pool, bathtub or wet basement.

**CAUTION: Temperature**

- Surfaces of the device may become hot during operation.
- Do not install this device near any heat source such as radiators, stoves or other heat sources.
- Always allow enough ventilation space around the unit for air circulation.
- Do not cover circulation vents.

**CAUTION: Connecting & Mounting**

- Never connect the output of a power amplifier to this device.
- Place the unit on a rigid board or fix it to an appropriate rack.
- Use the device according to this manual only.

**CAUTION: Humidity**

- If this device is moved from a cold place to a warm room, condensation can occur inside the device. To avoid damaging the unit please allow it to reach room temperature before switching on.

**CE Conformity**

elysia GmbH, Ringstraße 82, 41334 Nettetal, Germany, declares with sole responsibility that this product complies with the following norms and directives:

- 2006/95/EG Low Voltage Directive (formerly 73/23/EWG or 93/68/EWG)
- 89/336/EWG EMC (Electromagnetic Compatibility) Directive
- DIN EN 55103-1 EMC of audio equipment - Emission
- DIN EN 55103-2 EMC of audio equipment - Immunity

This declaration becomes invalid by any unapproved modification of the device.

Nettetal, 01.06.2006 - Ruben Tilgner & Dominik Klaßen

## Dear friend of audio culture,

First of all, we would like to thank you sincerely for choosing the alpha compressor as your new mastering tool. From now on, you not only have the maximum possible audio quality in dynamics processing at your disposal, but also a complete arsenal of flexible and effective tools which will shift the fruit of your work to a new level.

In short: You have opted for a product designed without the slightest compromise!

Please take a little time to read this manual thoroughly, as it will help you to entirely understand the enormous potentials and to really push the envelope. We have attached great importance to practical experience and fast results, which is also the reason for reserving the explanation of the technological excesses realized in the alpha compressor to our website.

The Basics chapter contains essential information for the fundamental comprehension and quick use of the specific functions and modules. After this, a couple of Scenarios are introduced in order to illuminate just some of the versatile applications of your new favorite compressor. If you want to go into further details, please have a look at the Reference chapter which elaborates on every particular parameter.

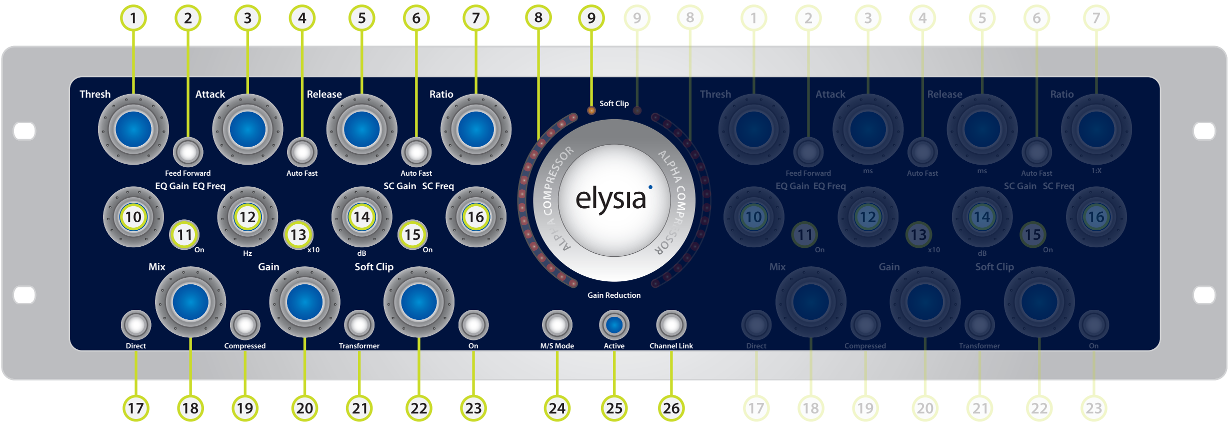
If you have further questions or comments, please do not hesitate to contact us – we enjoy being of your service. But for now, it is time to wish you lots of fun and pure audio pleasure with your alpha compressor.

Use the Force…

**the elysians**

## Controls

Both channels of the alpha compressor are absolutely identical regarding their electronic design. Therefore both sides of the front panel have exactly the same controls and switches. Continuative information on the specific functions can be found on the pages in brackets.




① **Threshold:** the operating point of the compressor. If the input level exceeds the value set with this controller, the compression process will start. (p. 26)

② **Feed Forward:** switches the junction of the sidechain alternatively behind (feedback) or in front (feed forward) of the actual compressor section. (p. 14)

③ **Attack:** the transient response of the compressor. It determines the time the alpha compressor needs to reach 10 dB of gain reduction. (p. 27)

④ **Auto Fast:** a semi-automation. This function shortens the attack time automatically on fast and loud signal impulses. (pp. 13, 27)

⑤ **Release:** the return phase of the compressor. It controls the period of time between when the input signal falls below the threshold and the compressor's return to unity gain. (p. 28)

⑥ **Auto Fast:** similar to the Auto Fast feature for the attack parameter, this function automatically shortens the release time and then returns to the set value. (pp. 13, 28)

⑦ **Ratio:** the relation between the input level and the output level. In feed forward mode, the actual ratios become significantly higher than the printed values. (p. 29)

⑧ **Gain Reduction:** the display for the gain reduction. Shows the amount of compression measured in dB as an optical support for the acoustic events. (p. 12)

⑨ **Soft Clip LED:** indicates the activity of the Soft Clip Limiter. This LED should only light up shortly from time to time in order to avoid audible distortion. (p. 19)

⑩ **EQ Gain:** the characteristic of the Niveau Filter. Between the mid and fully counter-clockwise position, bass is boosted and treble is cut (vice versa in the other direction). (pp. 17, 30)

11 **EQ On:** activates the Niveau Filter. In the signal path, this special EQ is placed after the compressor section, thus it will not influence this section's behavior. (p. 17)

12 **EQ Freq:** the center frequency of the Niveau Filter. Around this reference point, bass is boosted and treble is cut or vice versa. (pp. 17, 31)

13 **x10:** shifts the frequency range of the Niveau Filter. The printed values from 20 Hz to 2.0 kHz are multiplied by 10 to 200 Hz and 20 kHz. (pp. 17, 31)

14 **SC Gain:** influences the character of the sidechain filter. It can be blended from high pass to low pass with lots of expedient interim values. (pp. 16, 31)

15 **SC On:** activates the sidechain filter which allows frequency-dependent shaping of the compression process. (p. 16)

16 **SC Freq:** sets the highest/lowest frequency which – dependent of the SC Gain controller – you want the compressor to react to/not to react to. (pp. 16, 32)

17 **Direct:** makes the direct signal routed directly from the input stage audible in its respective channel or mutes it when inactive. (p. 18)

18 **Mix:** if both the Direct and Compressed knobs are active in their respective channel, the Mix controller blends between them for onboard parallel compression. (p. 18, 33)

19 **Compressed:** makes the compressed and/or filtered signal audible in its respective channel or mutes it when inactive. (p.18)

20 **Gain:** compensates for the level reduction caused by the compression process. The controller offers up to 12 dB of amplification. (p. 9)

21 **Transformer:** switches an additional transformer into the signal path as an additional means of sound shaping. (p. 14)

22 **Soft Clip:** helps to limit short, loud transients to subsequent A/D converters from clipping by rounding the signal peaks. (pp. 19, 33)

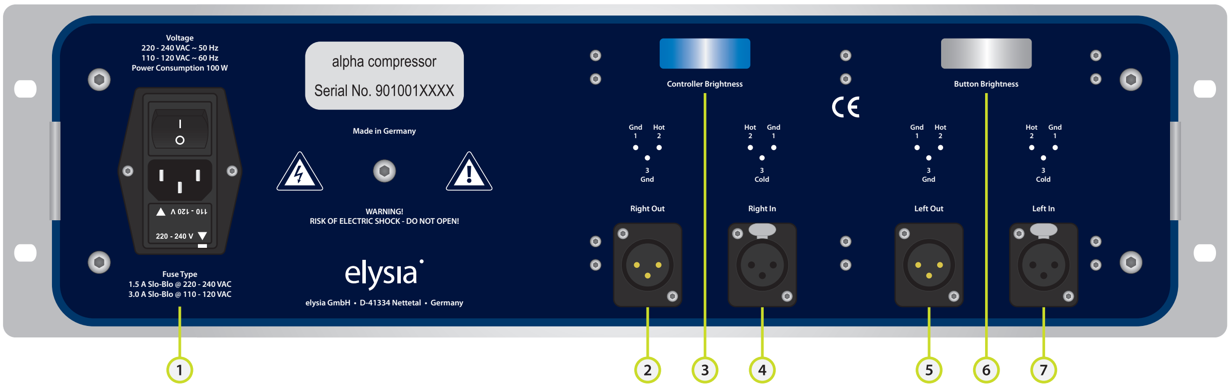
23 **Soft Clip On:** activates the Soft Clip limiter. The limiter modules for both channels should always be activated or deactivated at the same time. (p. 19)

24 **M/S Mode:** enables to process the middle and the side signals separately, and finally decodes both channels back to left and right/stereo. (p. 15)

25 **Active:** enables the processing modules of the alpha compressor. In deactivated state, the input is directly routed to the output by a hardwire bypass. (p. 10)

26 **Channel Link:** both channels can be operated in combination with the left control panel as master. *Note*: Filter, mix, gain and limiter stages will not be linked in this mode. (p. 11)

## Connectors



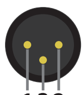

1 **Mains module**
This module combines the line cord connector, the on/off switch, the fuse holder with integrated 230/115 VAC voltage selector and a line filter for providing the transformer with clean current. *Note*: Some export version have a fixed voltage of e.g. 100 or 115 VAC and cannot operate at 230 VAC.

**WARNING: High voltage**
Make sure to disconnect the line cord before replacing eventually blown fuses or changing the operating voltage of the unit!

In order to change the operating voltage, the fuse holder has to be taken out and re-inserted so that the desired voltage can be read correctly (and is not standing upside down). The white triangular arrow that belongs to the chosen voltage setting points at the small rectangular mark on the mains module.

**WARNING: Fuses**
Always make sure to use the correct fuses for the chosen voltage: **230 VAC 1.5 A Slo-Blo** or **115 VAC 3.0 A Slo-Blo**. Incorrect or missing fuses are dangerous safety hazards for both the unit and yourself!

2 **Audio outputs (+4 dBu)**
5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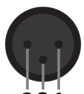
| | | | |
|---|---|---|---|
| Pin assignment balanced: | 1 ground | 2 hot (+) | 3 ground |
| Pin assignment unbalanced: | 1 ground | 2 hot (+) | 3 idle |

4 **Audio inputs (+4 dBu)**
7


231

| | | | |
|---|---|---|---|
| Pin assignment balanced: | 1 ground | 2 hot (+) | 3 cold (-) |
| Pin assignment unbalanced: | 1 ground | 2 hot (+) | 3 ground |

*Note:* If a device that is placed before the alpha compressor in the signal chain has an unbalanced output stage, a complete muting can eventually occur when the compressor is activated. If this happens, please follow the advice on page 34.

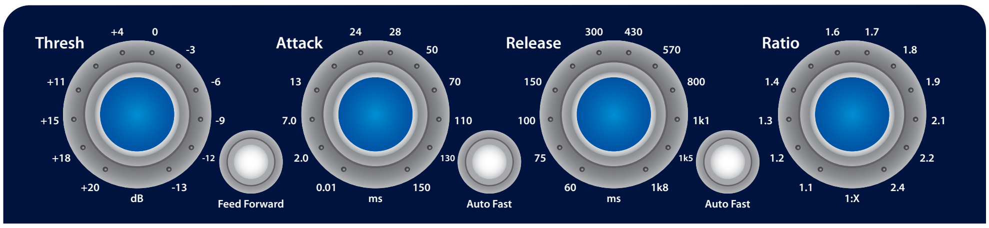
3 **LED brightness dimmers**
6

| | |
|---|---|
| Controller brightness: | light intensity of blue LEDs |
| Button brightness: | light intensity of white LEDs + logo disc |

## Getting Started

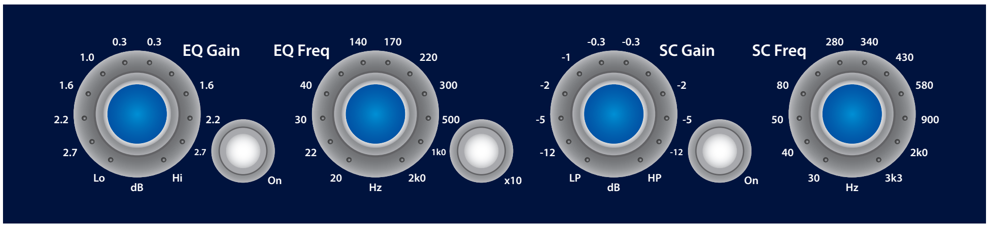
Never fear! With its many options and quite a lot of controllers the alpha compressor might look a little tricky at first sight. In practical experience, however, it is a clearly structured and easy to use tool.

The control elements are divided into three layers, whereas the left and the right channel sides are completely identical.



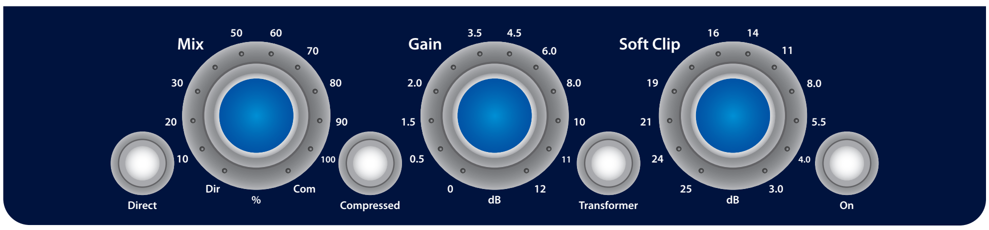

### Compressor section

The upper array contains the dynamic section with the classic parameters threshold, attack, release and ratio. Furthermore, there are some interesting special functions like feed forward and Auto Fast that will be explained later.




### Filter section

Firstly, the middle array features the audio filter, which enables you to perform subtle changes on the overall tonal character. Secondly, it consists of the sidechain filter which allows frequency-dependent compression when it is switched into the detector path.




### Level section

The lower array starts with the mix controller that allows blending between the unprocessed and the compressed, filtered and amplified signals. Next in line is the gain controller for making up the gain of the reduced signal. Last comes the Soft Clip limiter which allows catching fast transients.

In the middle of the unit, right below the gain reduction meters, there are three buttons to set the distinct operating modes of the alpha compressor:


### M/S Mode

If this button is pressed, the alpha compressor will work in M/S mode in which the controls on the left side affect the middle channel and those on the right side belong to the side channel. Otherwise the unit will work in stereo mode.

### Active

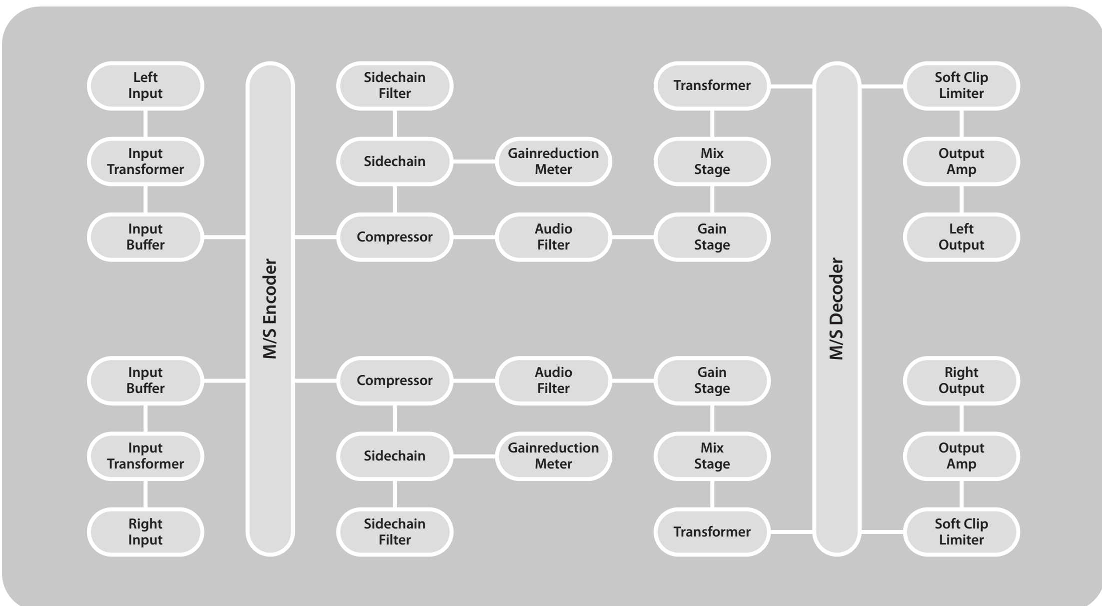
If this button is pressed (LED glows), the incoming signals will be processed by the alpha compressor. Otherwise there is a hard bypass, whereas the gain reduction meters will still stay active.

### Channel Link

If this button is pressed, the left upper array will act as a master for the dynamic sections of both channels. All other parameters (filter, mix, gain, limiter) are not linked, though, and therefore have to be set manually. In this mode the gain reduction is identical for both channels.

## Signal Flowchart

The schematic shows the signal flow from the input through the specific modules to the output. M/S matrix, sidechain filters, audio filters, mix stages, transformers and Soft Clip limiters can be optionally switched into the signal path via relays, which is also true of the Auto Fast and feed forward functions.




## Link Mode

In link mode, both channels of the alpha compressor can be coupled and operated with the controllers and switches of the left control panel. Please consider the following important notes:

Both sides of the alpha compressor utilize exactly identical electronic modules that could even be interchanged with each other in principle. This grants a high equality of signal processing in the left and right channel, which is especially important in stereo mode. Another result is the excellent isolation of both channels from each other, and the audio quality will not decline because there are no lossy to and fro signal routings in the circuitry.

As an effect of this design approach, only the *control functions* of the compressor (marked green in the figure) are combined in link mode. The left control panel becomes the master, that is the settings for the left channel are transferred to the right channel – therefore the settings of the particular controllers on the right side are not relevant in link mode.

At the same time this means that the *audio functions* of the compressor (marked orange) are not combined. In linked stereo mode please pay attention to set the controllers and switches for both EQ, mix and gain sections identically. Otherwise an audible drift between the left and the right channel might occur.

In linked M/S mode, however, you can set the EQ, mix and gain stages to different values to achieve certain effects. For example, you can use dissimilar settings of the gain controllers to alter the stereo width (more information on page 24) or you can employ the EQ stages for mid and side signals separately.

In the signal path, the Soft Clip limiters are located after the M/S decoder. This means that the limiters are completely independent from the operation mode of the alpha compressor – they always work in stereo. The limiters are also not affected by the status of the link function and therefore have to be set at the same values every time they are engaged.


## Stereo Compression

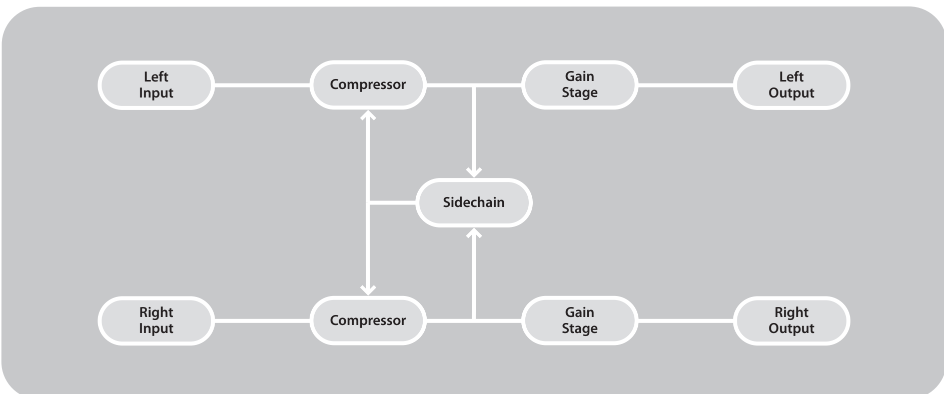
In order to get a first impression of the sonic qualities and the control characteristics of the alpha compressor, only the controllers highlighted in the figure are used. The functions that are currently not needed should be deactivated (related LEDs do not glow). Et voilà: We have a classic linked stereo compressor!

A track with a broad dynamic range is the best material for a first test. Just start playing with the threshold, attack, release and ratio parameters of the left channel in order to get to know the control behavior of the compressor. Allow yourself to take some time for this, as the understanding of the basic control characteristics is the most important condition for perfect results.

Keep having a look at the gain reduction meters from time to time. No doubt: the ear is the decisive benchmark. Just consider the optical information as a supporting means to evaluate the compression process (fast/slow, strong/weak, etc.).

The levels of the compressed signals can be made up by adding an appropriate amount of amplification with the gain controllers. Since the link function will only affect the control voltage of the compressor, the gain controllers of both channels should be set to the same values in stereo link mode. For instant A/B comparisons just push the active button.

*Note*: If the threshold controller produces too little gain reduction when turned hard right or too much of it when turned hard left, please go to page 26 and see how to adjust the internal threshold.



## Auto Fast (Attack)

The Auto Fast function causes an automatic shortening of the attack time on fast and loud signal peaks. This effect is easy to comprehend when compressing a preferably dynamic drum loop, for example.

Please set the following values as a starting position:

- Attack: 50 ms
- Release: 300 ms
- Ratio: 1:1.8
- Threshold: Set to a position that results in approx. 3-4 dB of gain reduction

Because of the quite long attack time the snare and bass drum will be processed very moderately. If now Auto Fast is engaged, the compressor will react to the fast and loud peaks, resulting in a noticeably higher amount of gain reduction.

And here is the trick: Directly after this process the attack time will return to the value that was originally set, and on ‚slower' signals the original value will not be changed at all. The compressor will only become very fast in case it is really needed!

## Auto Fast (Release)

The same example is suitable to use the Auto Fast function for the release parameter, too.

The following values are useful to get started:

- Attack: 30 ms + Auto Fast
- Release: 300 ms
- Ratio: 1:1.8
- Threshold: Set to a position that results in approx. 7-8 dB gain reduction

Now press the Auto Fast button of the release controller and keep your ears open as well as an eye on the meter: the area between 4-8 dB is reduced very fast now, whereas the set value of 300 ms stays valid between 0 and 4 dB.

*Note*: If you use very long release times, the effectiveness of this function will decrease. Very long attack times will also reduce the intensity of this circuit.

This function can be a great help especially for applications with difficult dynamic structures, as it counteracts the danger of clippings caused by too short values and the loss of loudness and pressure caused by too slow settings at the same time.

## Feed Forward

This function makes it possible to switch the junction of the sidechain alternatively behind (feedback) or in front (feed forward) of the actual compressor section. This has an enormous effect on the character of the compressor.

Again, our drum loop can be used for demonstration: While processing in feedback mode is smooth and even, switching into feed forward mode will result in a clearly more aggressive and harder kind of compression.

From the technical point of view this function mainly influences the characteristic curve of the ratio value. In feedback mode it goes up to a moderate Ratio of 1:2.5. In contrast, the feed forward mode provides much higher values that also allow limiter settings and even negative ratios (i.e. loud signals will be reduced even more). Thus, small changes in dynamics can generate a high amount of gain reduction.

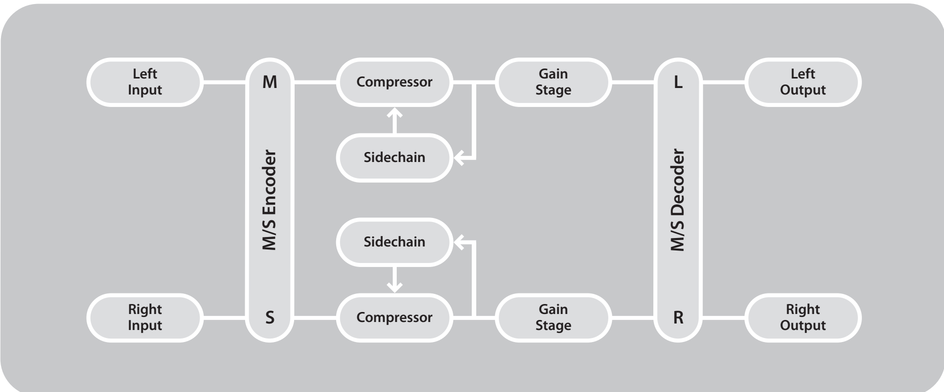
The values of the attack controller will also change noticeably in feed forward mode, as they will become almost twice as high as the value shown on the scale.

## Transformer

Each of the two channels features an additional transformer that can be switched into the signal chain after the mix stage. In principle, these are classic output transformers, but here they are not used for balancing and galvanic isolation, but as an additional means of sound shaping.

If you like to add that certain amount of ‚iron' to your sound, just push the transformer button and there you are. Because of the mastering approach of the alpha compressor this feature is much more a subtle audio shaping feature than a glaring sound effect.

In stereo mode, the buttons for both channels should be activated or deactivated at the same time. Depending on the source material and personal taste, you can, however, add the transformer sound to single channels when working in M/S mode.




## M/S Matrix

One of the most powerful features of the alpha compressor is its switchable M/S matrix. By activating it, the incoming stereo signal is encoded into a middle channel (mono) and a side channel which contains the stereo parts of the left and right channel (but not the common mono part).

This enables you to process the middle and the side signals separately, and finally both channels are decoded back to left and right (stereo). A practical example helps to understand the general idea.

The ideal song for this would be the following: Important elements like voice, drums and bass are placed in the middle whereas the remaining instruments are mixed stereophonically.

Switch the alpha compressor into M/S mode and leave the link function deactivated for now. The left side of the compressor now controls the middle channel and the right side the side channel. By using the compressed switches, you can listen to both channels separately, so that you will be able to hear what is instantly happening in the particular channel.

Now you can process the mono and side parts separately from each other. Depending on the mix, sometimes it is only the middle channel that needs compression whereas the side channel stays unprocessed. Other mixes have a lot of stereo information; in these cases both middle and side channels are processed, the settings of which can be completely different from each other.

At the beginning, both gain controllers should be set at the same values, but this can change in the progress of optimization by all means. You can use different settings of the gain controllers in M/S mode for specific changes of the stereo width, for example.

Of course the link function also works in M/S mode. In this case, signals in the M and S channels are reduced evenly so that the result of the compression is the same as if working in stereo mode. However, you will still be able to adjust the level proportion between M and S with the gain controllers.


## Sidechain Filter

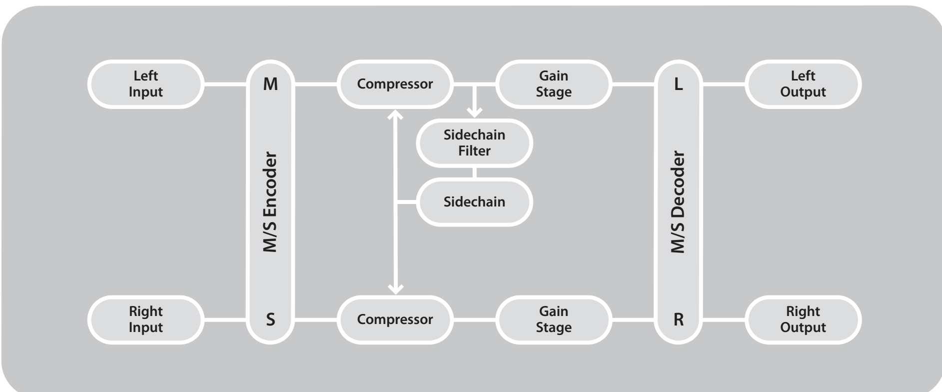
The sidechain filter allows frequency-dependent shaping of the compression process by giving specific frequency areas a stronger or weaker influence on the detection circuit.

The linked M/S mode is most suitable to demonstrate the effect: Since the largest amount of signal energy can be found in the middle as a general rule (bass drum etc.), the filter will have its greatest impact here. A track with a balanced proportion of bass, middle and treble frequencies should be chosen to give this function a try.

If the SC gain controller is set to HP (High Pass), the filter will act like a 6 dB high pass and the reaction of the compressor on bass frequencies decreases. The setting LP (Low Pass) turns the filter into a 6 dB low pass and the compressor reacts primarily on low frequencies. Again, the gain reduction meter is a reliable aid to evaluate the distinct modes of operation.

The combination of sidechain filters, M/S matrix and different attack settings enables you to make very selective changes and – depending on the source material – even to process single instruments or voices in finished mixes. Details on that can be found in the Scenarios chapter.

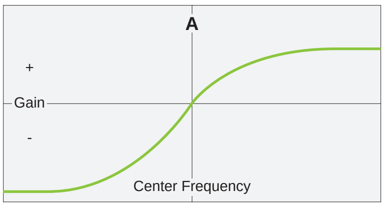
*Note*: When working in linked stereo mode, the sidechain filters of both channels should be set to identical values. In linked M/S mode, however, only the sidechain filter of the middle channel is active, and in unlinked M/S mode the SC filter of the side channel is available, too.

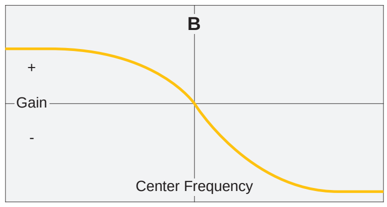


## Audio Filter

The audio filter is placed behind the compressor in order to avoid unwanted influences on the detection circuit of the dynamic section.

If the EQ gain controller is turned hard right, the signal rates above the selected center frequency will be boosted by 3 dB and the part below the center frequency will be cut by 5 dB [Fig. A]. Turning the controller hard left reverses the effect: Below the center frequency there is a boost and above it there is a cut [Fig. B].

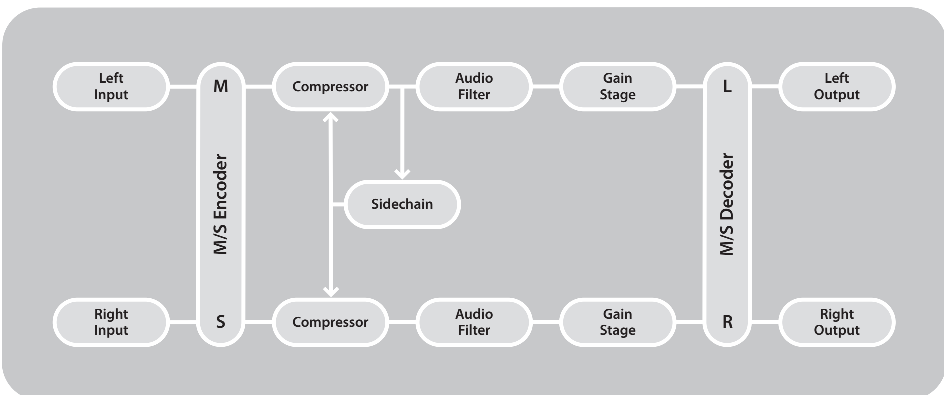



The EQ frequency controller comprehends an array from 20 Hz - 2 kHz which can be shifted to 200 Hz - 20 kHz by pushing the x10 button.

In linked M/S mode as explained above it is possible to make different settings with the audio filters in the M and the S band. Again, the results can be listened to separately by using the compressed switches for each channel. However, in stereo mode, both audio filters should be set to the same values in order to avoid unwanted shifts in the stereo spectrum.

*Note*: Depending on the settings of the audio filters it might be necessary to adapt the volume with the gain controllers.




## Mix Controller

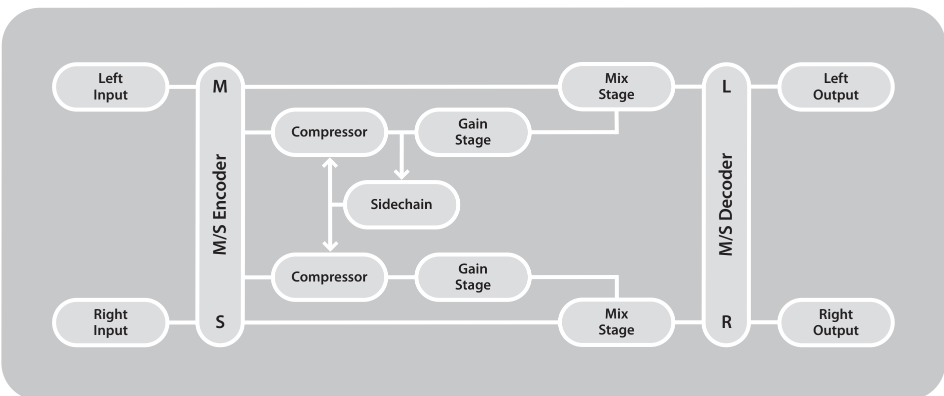
The mix controller makes it possible to cross-fade between the unprocessed and the compressed/ filtered signals. This allows parallel compression right in the alpha compressor and renders additional routings unnecessary in favor of a better signal quality.

Now you can use even extreme compression settings without killing the track by winning the loudness war. By mixing just a part of the compressed signal to the original, the major portion of the initial dynamic structure remains.

The control logic of this feature is also worth mentioning. If only the compressed button is pushed, you will hear the compressed signal only. Otherwise, if only the direct button is pushed, only the unprocessed signal routed from before the compressor section will be audible. If both buttons are pushed, the mix controller will become active, and if none of the buttons is pushed, the channel will be muted.

In practice you can switch between unprocessed, compressed or mixed signals very fast without having to change the position of the mix controller.

In this way, the left and right channels (stereo mode) respectively the middle and side channels (M/S mode) can be listened to separately – here you again have the choice between the original, the compressed or the mixed signal. To mute the other channel, just deactivate the associated direct and compressed buttons.



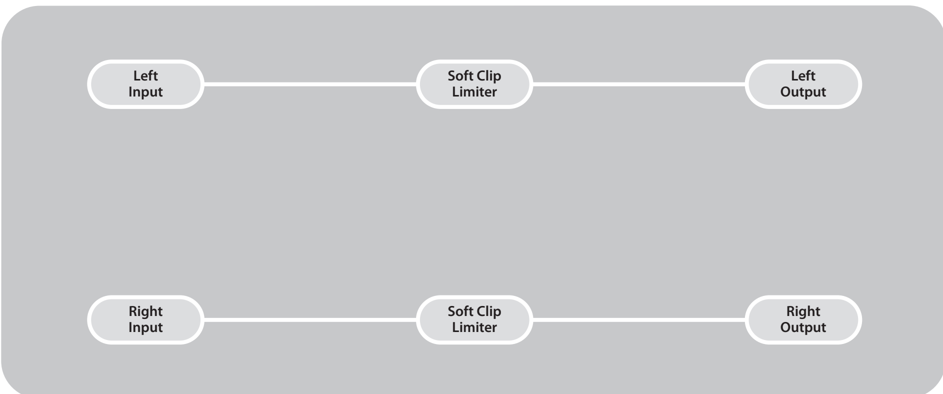

## Soft Clip Limiter

The Soft Clipper was designed to limit short, loud transients to prevent subsequent A/D converters from clipping. However, the circuit does not behave like a classic brick wall limiter, but more like an analog tape machine driven into saturation: it rounds the peaks instead of making hard cuts.

The Soft Clip modules are placed directly in front of the output stages. Their settings are not influenced by the mix and gain controllers. This is also the case in M/S mode, because the limiters are located after the M/S decoder and therefore are always assigned to the left and right stereo channels. Note that the channel link does not affect the circuit either – because of this the settings of the Soft Clip controllers for both channels should always be identical!

If the alpha compressor is the last unit in the signal path before the A/D converter, it is only once that the setting of the Soft Clip limiter has to be adapted to the converter. It is best to use a fast, sample-accurate PPM meter to find the right setting, because it is an accurate method of benchmarking the resulting change in level.

The Soft Clip LEDs located on top of the gain reduction meters should only light up shortly, as noticeable distortion appears if you drive this circuit too hard. Particularly with regard to acoustic and classical music, this feature should be used with care and moderate settings only.




## Stereo Linked

This setting is especially useful for tracks in which the instruments are not placed straight in the middle of the stereo spectrum – when a recording has been made by using classic stereo miking like in Jazz, for example. In order to come up to such styles of music, one should make sure to create an unobtrusive and smooth kind of compression. This can be achieved by setting a long attack time and a ratio value in the lower control range, at which only a very few dB of gain reduction are required. If necessary, you can use the mix controllers to apply the effect of the compressor in even finer doses (both direct and compressed buttons have to be activated for this). Moreover, you can kick in the audio filters to make additional sonic corrections. *Note*: Look after setting the mix, gain and filter controllers at the same values when working in stereo mode. The link function only affects the control mode of the actual compressor section. Of course the linked stereo mode can also be used for many other styles of music, but then you will often be able to achieve even better results in M/S mode.


## M/S Linked

The function of the compressor in linked M/S mode is quite similar to its function in linked stereo mode, and the resulting gain reduction in the middle and the side channel is exactly the same. An attack time of 30-40 ms ensures that transients from percussion instruments are hardly compressed, while the release time of 250 ms lets the compressor decrease its gain reduction fast enough. Variations of the release time are especially effective in controlling the amount of loudness. Here, activating the Auto Fast function should be preferred to setting a very fast release time. The generated gain reduction is primarily affected by matching the threshold and ratio parameters. A ratio of 1:1.7 has often proved to be a good start. But what is the essential advantage of the linked M/S mode compared to the linked stereo mode? The levels of M and S can be set differently, which makes it very easy to perform changes of the stereo width (Mix too narrow? More gain to the side channel. Not enough punch? Add gain to the middle). Furthermore, a left/right drift caused by possible unequal settings of the gain controllers is impossible in this mode.


## M/S Unlinked

The true strengths of an M/S compressor will not be revealed until it is switched into unlinked M/S mode. Depending on the mix and the style of music, the dynamics and loudness in the middle and side channel can be completely different from each other. Take a heavily pumping bass drum right in the middle and 100 % stable strings in the sides, for example – just try this with any other compressor ;-) Thus it can make perfect sense to set strongly differing values of threshold, attack, release and ratio for the middle and the sides. In order to achieve a better evaluation of the results, it is advisable to listen to the processed channel solo for a start. Please remember to deactivate the direct and compressed buttons of the other channel for this. Using the audio filters, which can also be set completely different from each other now, can help to additionally enhance the tonal balance of a mix. Once you have discovered the potentials of the M/S mode, lots of formerly difficult mixes can be controlled much better, resulting in audibly better effects.


## Upward Leveling

Upward Leveling – this describes the possibility to make only the low signal parts louder and leave the louder changes in dynamics unprocessed. The settings shown above are suitable for classical music, for example, where ‚typical' compression is objectionable. The compressor works in linked stereo mode, the attack time is set at maximum and the release time is at 150 ms. Under these conditions the alpha compressor will function as a leveler. In feed forward mode the attack time is twice as long compared to feedback mode and therefore amounts to 300 ms. The ratio characteristic corresponds to a limiter, and the make up gain is about 6 dB. The result of these settings: Quieter passages that do not produce any gain reduction will be boosted by 6 dB, whereas loud signals will be limited by the compressor. Now the special trick is to set the mix controller to 50 %. By adding the original signal the initial dynamics will be saved, and the bottom line is that only the quieter signals will be turned up.


## M/S Leveling

As with Upward Leveling, long attack times are the secret of M/S Leveling. Here the compressor reacts primarily to the complete signal energy and less to single rhythmic accents. If this principle is now used on the side and the middle channel separately, a stereo mix will become more compact in an unobtrusive way and everything will sound a little more vivid. This method is especially interesting to enhance rock and pop tracks in a discreet but effective way. Here the feed forward mode is active again. The ratio is set at about 1:1.5 and the gain controller at about 3 dB. The threshold is adapted to a value that results in approx. 3-4 dB.


## Groove Compression

Sometimes it would be great to be able to just emphasize the rhythmical elements of a mix in order to bring out the groove with the aim of giving the track the right kick. The figure shows how the alpha compressor can help you to cope with this challenge. The bass drum is used as a trigger for the compressor; therefore the sidechain filter is of special importance in this application. It is set to low pass, with a center frequency of approx. 70 Hz. Now the threshold and ratio controllers are used to set the intensity of the gain reduction, and the groove will be controlled by the release parameter, which has merely to be adapted to the speed of the track. Depending on the song and the style of music, the gain reduction can be higher by all means; usually 5-6 dB will do very good. The linked M/S mode makes sense in this application, because the influence of the bass drum on the control process is enhanced as well as the resulting sonic effect.


## Vocal Down

If you have a track in which the vocals are too loud, the compressor can be set in a way that it mainly reacts to the lead vocals placed in the middle, resulting in a better integration into the mix. For this purpose, switch it into the unlinked M/S mode. Again, the specialty here lies in combining the sidechain filters with suitable attack parameters. The SC filter of the middle channel is set to high pass at a center frequency of about 270 Hz with the effect that the compressor hardly reacts to bass impulses anymore. The attack time lies between 70-100 ms. Thus, transient-like signals like a snare drum also only have a minor influence on the compression process. The release time is set at about 100 ms so that the compressor can return to its idle status fast enough. In the side band, both the controller for the threshold and the ratio parameters should be turned hard left if you do not wish any further processing there. An equal setting of the middle and side channel gain controllers is also advised.


## DeEssing

A variation of the Vocal Down scenario is DeEssing which is used for reducing unwanted ess and hiss sounds. The sidechain filter of the middle channel is set to approx. 2 kHz, which is even higher than in the previous application, so that only the high frequencies can influence the detection circuit. At about 100 ms, the attack time is also quite long and the release time is set to fast 50 ms. Now threshold and ratio are used to set the trigger point and the intensity of the processing. Again, this application is run in unlinked M/S mode to assure that only the middle channel is processed (but of course you can still additionally process the sides separately if needed).


## Parallel Compression

If it should (or must) become really loud, you can also use the alpha compressor to drive a track against the wall – but without the negative effects this usually entails. For this procedure it is preferable to work in the classic linked stereo mode. The time parameters are set very fast here; especially the attack time at only 10 ųS. The release parameter is set at 150 ms, while the corresponding Auto Fast function is also activated and the ratio is at maximum. With these settings everything will be compressed very intensely – even fast transients and impulses, resulting in an intense limitation of the output peak levels. Now the mix controller comes into play: If the just created ‚flat' signal is mixed to the original signal in moderate rates, it is possible to achieve an increase in loudness while still saving a great part of the initial dynamics.


## Stereo Enhancer

Even when there is nothing to compress from time to time, your productions can still benefit from the alpha compressor, as the combination of the M/S matrix and the audio filters makes a very nice tool for influencing the stereo spectrum. For this purpose the compressor is switched into M/S mode and the audio filter of the side channel adds 1.0-1.5 dB at a center frequency of approx. 1 kHz. The gain controllers of both channels are at 0 dB. These settings cause a boost of all frequencies above 1 kHz in the side channel, which intensifies the spatial perception as the room reflections become more distinct. The lower frequencies, however, are not of that great importance for spatial perception. Of course simply increasing the level in the side channel can be used to achieve a (broadband) change in the stereo spectrum, too, but this variant offers much more differentiated options because it is frequency-dependent.


## LoFi Compression

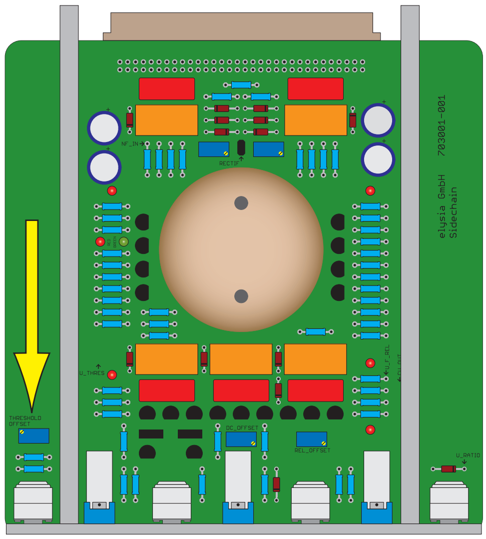
Can the alpha compressor also sound really bad? Of course not, but sometimes it just loves being, well … different. In this LoFi Compression scenario, the use of the audio filters is the crucial point. By reducing the high frequencies from 4 kHz on, everything sounds a little darker, while, at the same time, the bottom experiences a great degree of additional punch. The chosen time parameters result in loud and twangy compression effects. The high ratio settings create a great amount of compression, so that gain reduction values of 8-12 dB are absolutely normal. In this application the mix controllers give you the option to blend this extreme sound with the original at will.


## Extreme Settings

Even though the main focus of the alpha compressor lies on mastering applications, you can also use it to find interesting and sometimes extreme settings for single channels or subgroups. Here is just one of many examples. It is especially suitable for processing drum tracks effectively: The time parameters with the additionally activated Auto Fast functions are very fast. The high ratio in feed forward mode causes an extreme characteristic curve so that loud signals will be reduced significantly. By doing so, very high gain reduction values of 20 dB and more can occur by all means. The spatial elements now come into the foreground more articulately – it somehow sounds as if the microphones had been placed a little bit further away during recording.

## Threshold Offset

The analog audio level of the alpha compressor depends on the studio environment it is used in. In the majority of cases it depends on the D/A converter used, because the output stages of the converter determine the analog level. As there is no binding norm for it, this level can be at +4 dB, +6 dB, +10 dBu, +12 dBu or even +15 dBu. To make sure that the threshold controller actually covers its complete adjustment range it can be adjusted to those different levels.



**When is an adjustment necessary?**

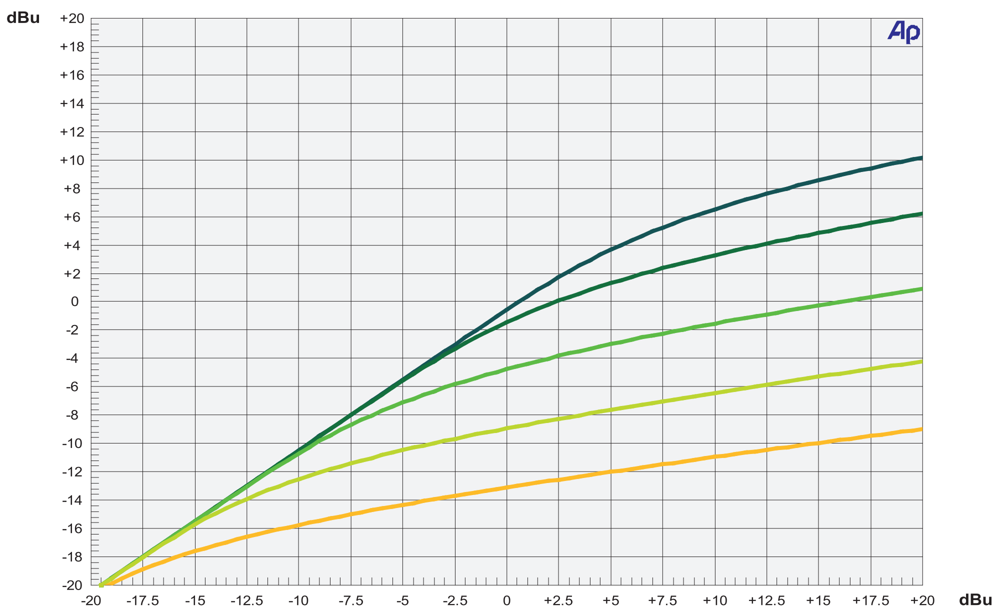
1. At low Ratio values (1:1.3), the threshold controller is almost completely turned clockwise and the gain reduction is still too low
2. At higher Ratio values (1:2.5), the threshold controller is turned completely counter-clockwise, but nevertheless gain reduction already sets in

In order to adjust the internal threshold offset, the first thing to do is to open the top cover with the adequate Torx screwdriver. Before the adjustment can be done the unit has to be powered up for at least 15 minutes to reach a stable working temperature. Please make sure to disable the M/S and link functions. Now send an audio signal into the compressor and set the attack to approx. 24 and the release to 300 ms.

On the upper circuit board there is a blue trimmer labeled THRESHOLD OFFSET placed directly behind the threshold potentiometer (yellow arrow in the figure). In the first case (too little gain reduction) the trimmer has to be turned clockwise to raise the amount of gain reduction. In the second case (too much gain reduction) the trimmer has to be turned counter-clockwise. In order to reassure that the adjustment range now really fits, it is recommended to repeat the above-mentioned test with the threshold controller in extreme left/right position. The same procedure should also be applied to the right channel, of course.

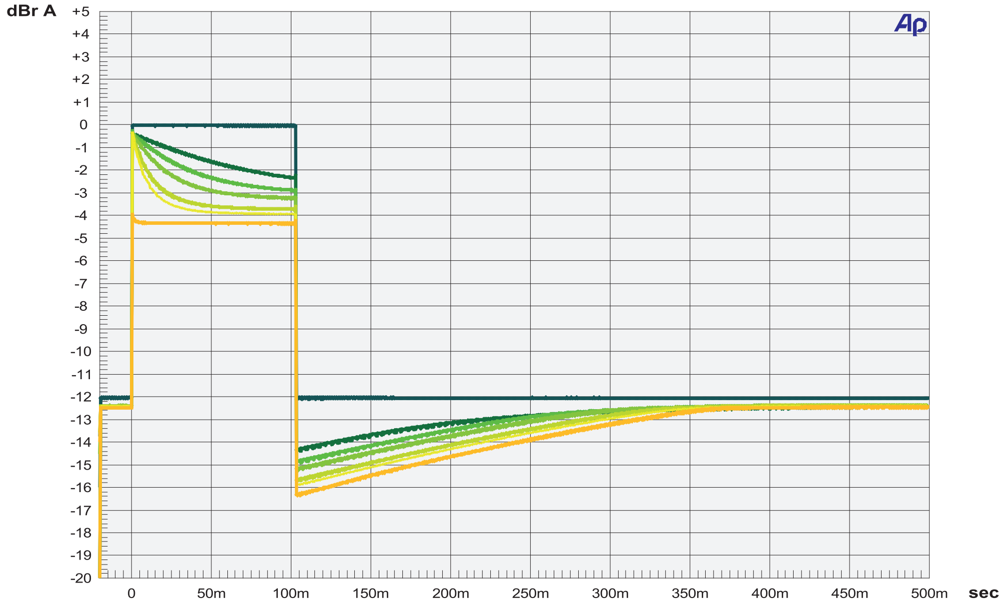
## Threshold

The diagram shows different threshold settings in feedback mode. The threshold and the ratio are always interdependent: On high ratios the threshold is usually set to lower values in order not to produce too much gain reduction, and on lower ratio settings the threshold controller is turned more to the right. The complete threshold range covers 33 dB which enables the user to make very sensitive settings in reasonable steps at any time.



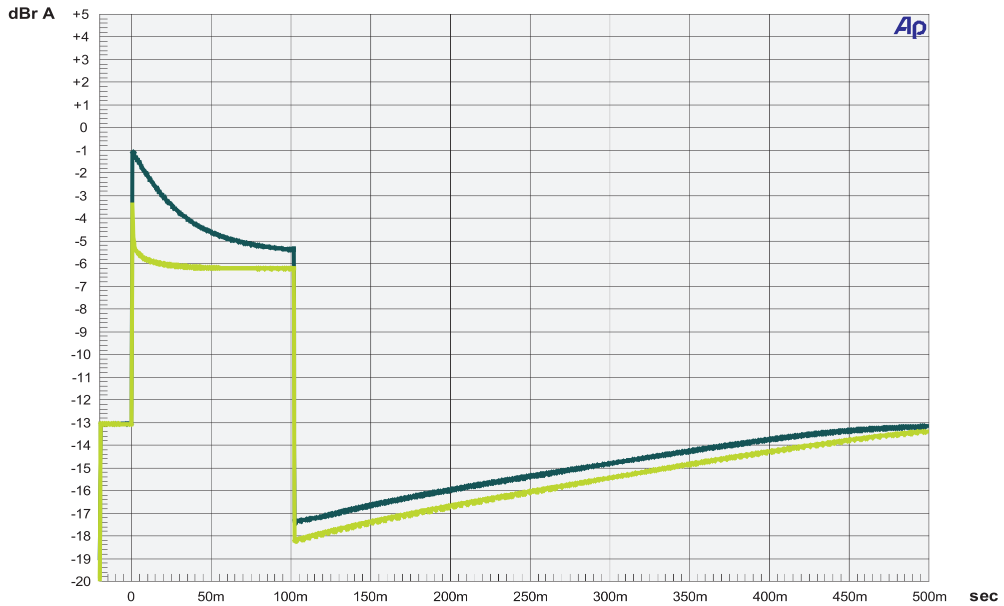
## Attack

Here a burst signal (a sine tone that is boosted by 12 dB for a period of 100 ms) is used to demonstrate the effectiveness of the attack controller. The different curves show how fast this level jump is reduced by 4 dB. On fast settings the reduction takes only a few milliseconds, on longer settings the reduction does not only take longer, but it is also smaller. Therefore the amount of gain reduction is always higher on fast attack settings than on longer settings.



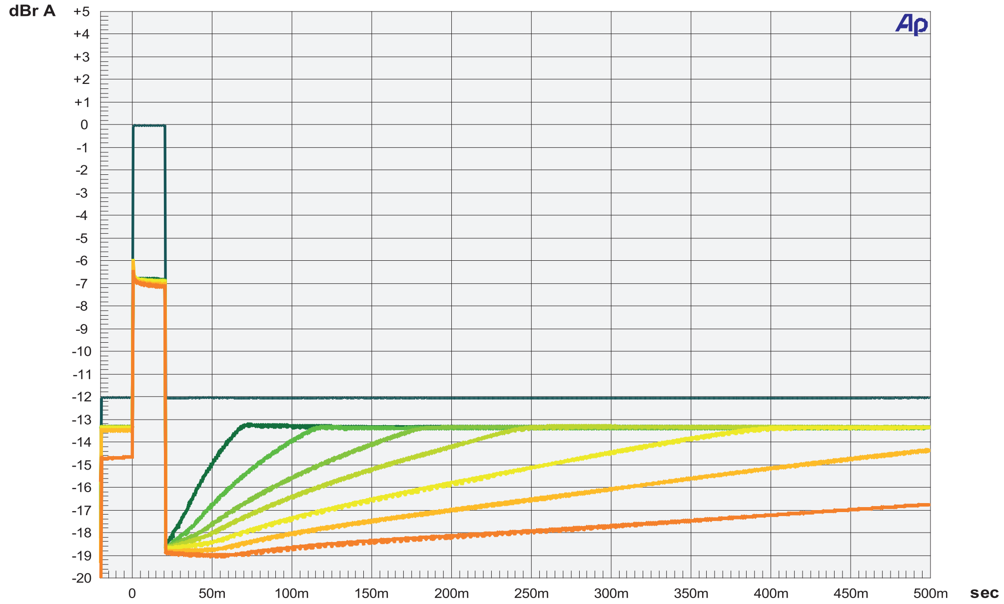

## Auto Fast Attack

Utilizing a burst signal again, this diagram shows a comparison between a normal attack time of 50 ms (dark curve) and the later added Auto Fast function (light curve). This reveals how fast this circuit reacts: Even the first waves of the burst are already reduced by 2.3 dB. The whole gain reduction now happens much faster, too. *Note*: This function starts working at -3 dB; on lower gain reduction values the changes in dynamics are too small to trigger it accurately.



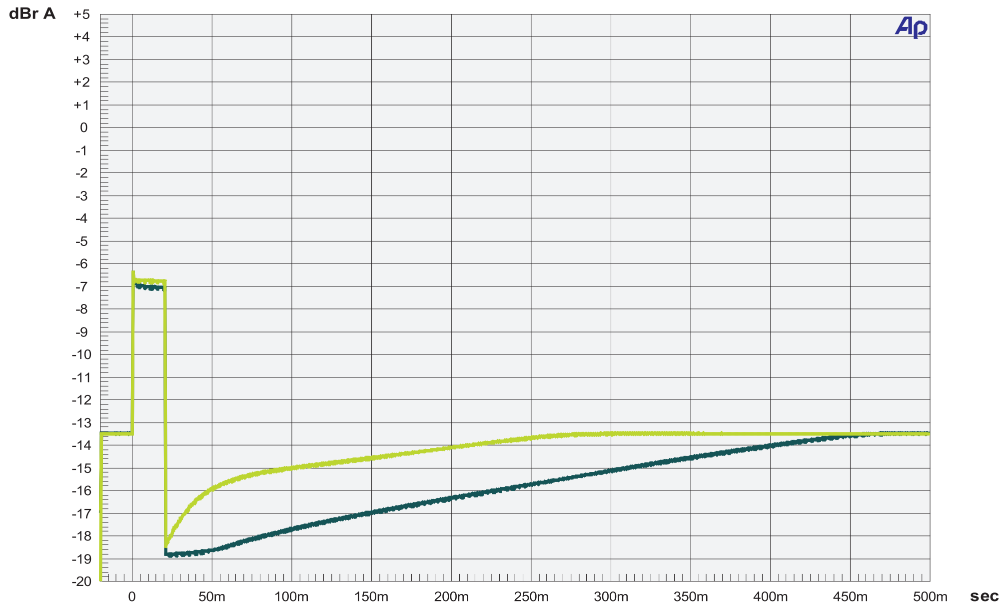
## Release

This figure is based on a very fast attack time; the gain reduction is about -7 dB. The burst lasts for 25 ms and then follows the release phase with time parameters of different length. The linear characteristic of the release curve is very easy to see here. Because of the utilized circuit, the attack time must always be added to the release time. The advantage is that in combination with long attack settings you can use even the fastest release settings without generating unwanted artifacts.



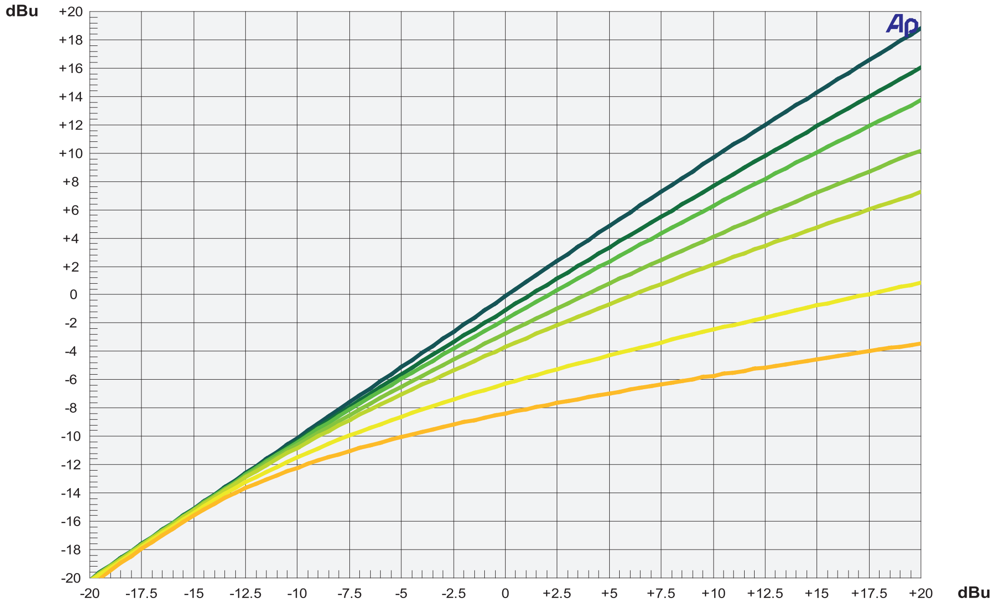

## Auto Fast Release

In this case the Auto Fast function was added to a release time of 400 ms (light curve). It is easy to see that directly after the burst the release time becomes very fast (only 20 ms) and then returns to its original setting. This has two effects: Firstly, the level in a period of 500 ms after the burst becomes 2.5 dB louder, and secondly the release phase is already completed after 250 ms. In practice this means an apparent increase in loudness without the distortion usually generated by constantly fast release times.



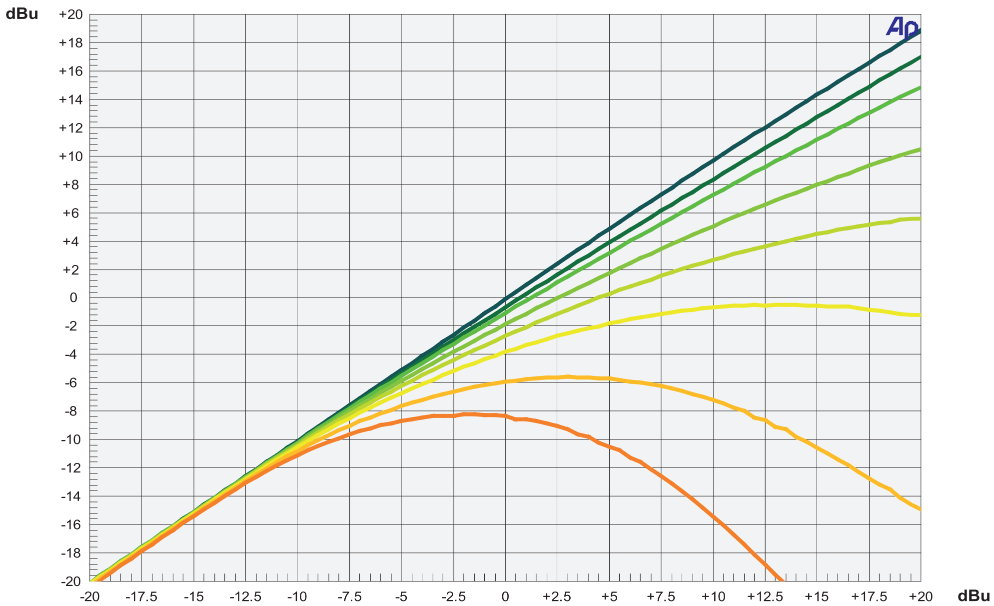
## Ratio Feedback

The alpha compressor works with a soft knee characteristic and increasing ratios at rising gain reduction values. The scale on the front panel is based on a measured gain reduction of 6 dB. Strictly speaking, the ratio values become noticeably higher on stronger reductions: If there is 17 dB of gain reduction the ratio value will shift from the printed 1:2.5 to actual 1:4.5 after all. Anyhow, the scale is optimized to the lower gain reduction settings as these are the ones most frequently used in mastering applications.



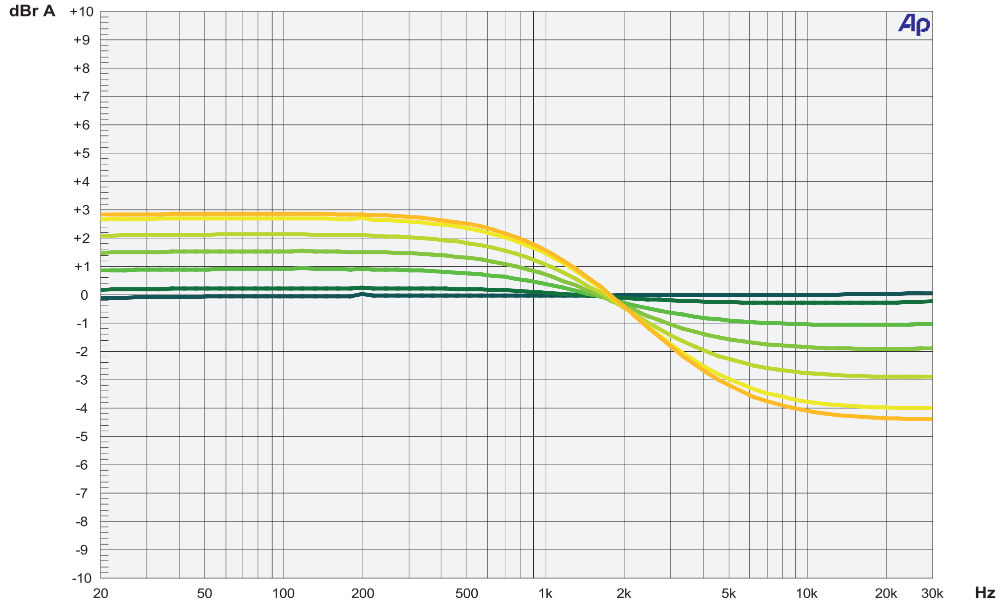

## Ratio Feed Forward

In feed forward mode the ratio characteristics differ strongly from those in feedback mode. Although they remain quite comparable up to a value of 1:1.5, the differences become extremely obvious at higher settings. The lower curve shows the radical effects: An input level of 0 dBu is reduced to -8.3 dBu, and an input level of +10 dBu is reduced to -15 dBu (!) after processing. Extreme settings like these can only be achieved with a feed forward compressor, and they can result in very wild effects.



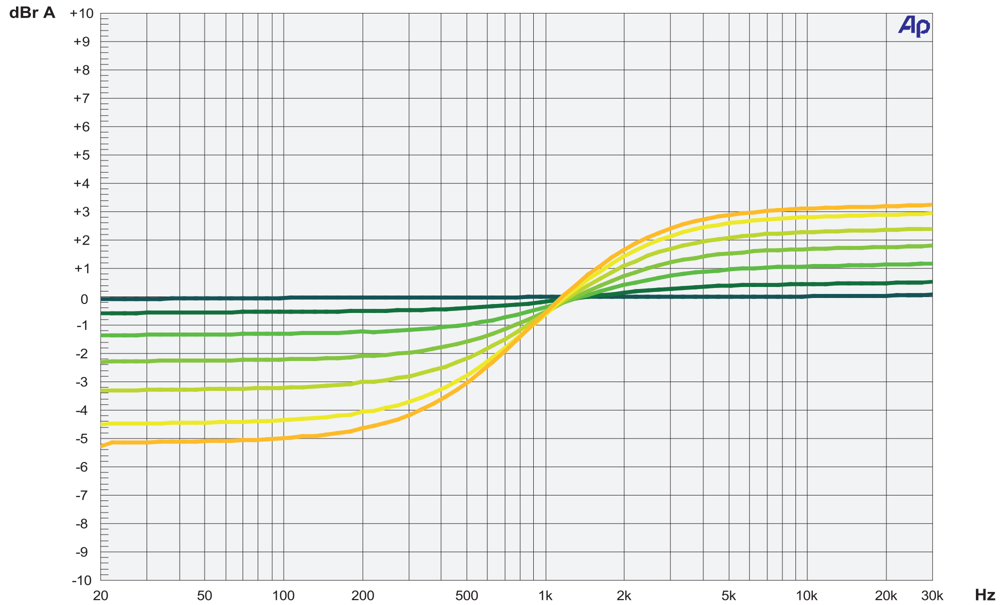
## EQ Gain Low

For this measurement the center frequency was set to 1.8 kHz and the EQ gain controller was set to different positions between 0 and full left (-3 dB). The diagram shows the effectiveness of the filter: The lower frequencies are boosted up to 3 dB, at the center frequency they cross zero, and the higher frequencies are cut up to 4.5 dB. Depending on the setting of the EQ gain the level might need some make up with the gain controller.



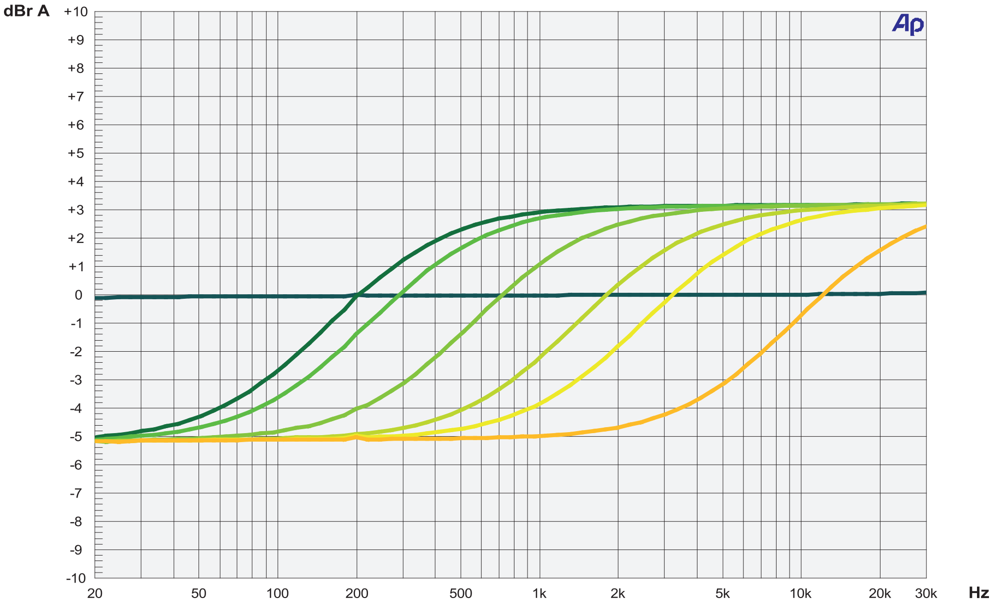

## EQ Gain High

Now let's have a look at the opposite scenario. In this figure the EQ gain controller was turned from 0 to full right at a center frequency of 1.1 kHz. Contrary to the previous example, the high frequencies are now boosted up to 3 dB and the lower frequencies are reduced by up to 5 dB. The fine-tuned interim values make it possible to apply very subtle sonic changes.



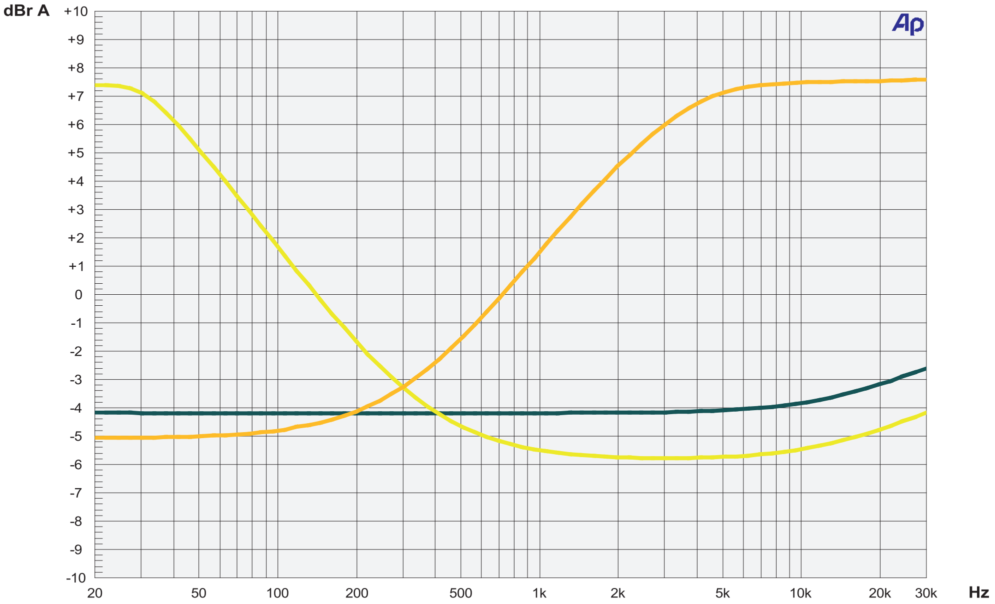
## EQ Frequency

This diagram shows a variation of center frequencies with the EQ gain controller turned fully clockwise (HI). The x10 button shifts the complete frequency area by a factor of 10. With settings in the border areas of the covered frequencies it is also possible to realize 'almost' shelving filters, but in these cases the level has to be adapted with the gain controllers.



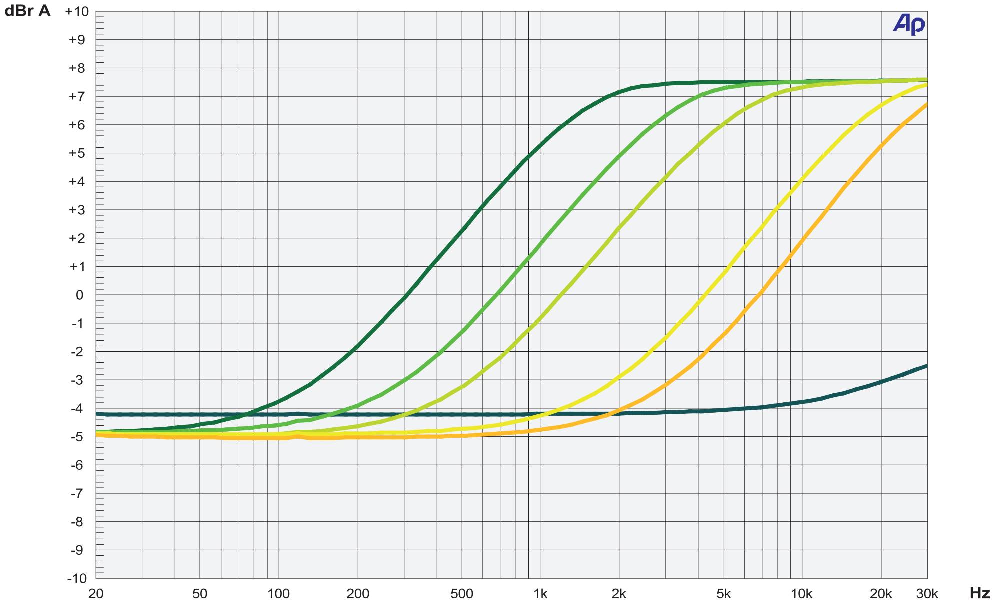

## SC Gain

The green curve shows an almost linear compression with a constant gain reduction across the complete frequency spectrum. If the SC gain controller is set to high pass (HP – yellow curve), however, the low frequencies will not influence the detection circuit and the gain reduction sets in only on higher frequencies. In low pass mode (LP – orange curve) just the low frequencies are fully processed and the intensity of the gain reduction decreases with rising frequencies. Of course there are a lot of useful interim values between the linear and the extreme positions.



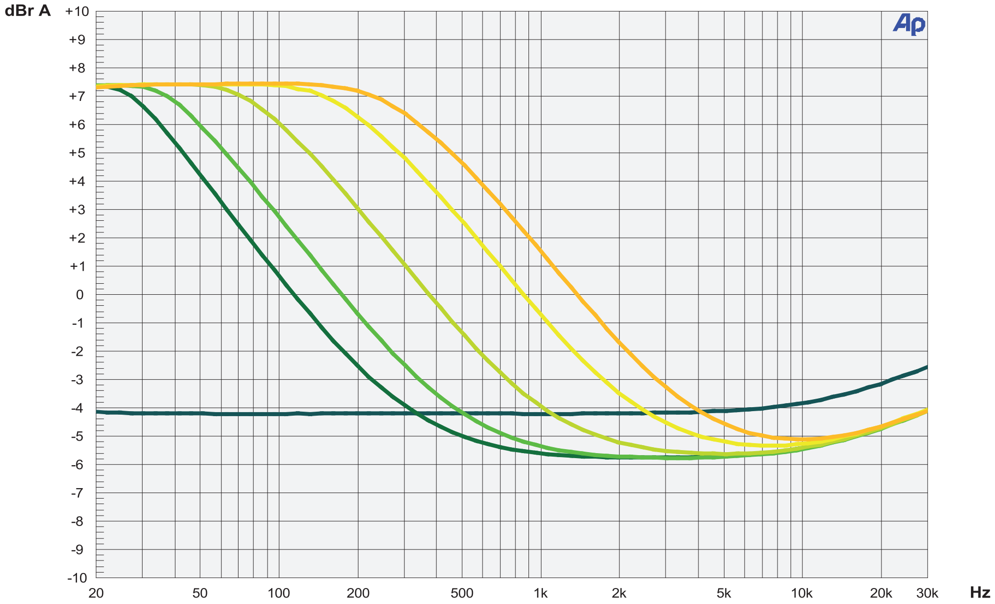
## SC Low Pass

This diagram shows some settings of the sidechain frequency controller (SC Freq) in low pass mode (LP). If the filter is set at 100 Hz for example, full gain reduction will still occur at these 100 Hz and the high frequencies will have lesser influence the higher they get. Therefore in this mode of operation the filter should always be set to the highest frequency you just about want the compressor to react to.




## SC High Pass

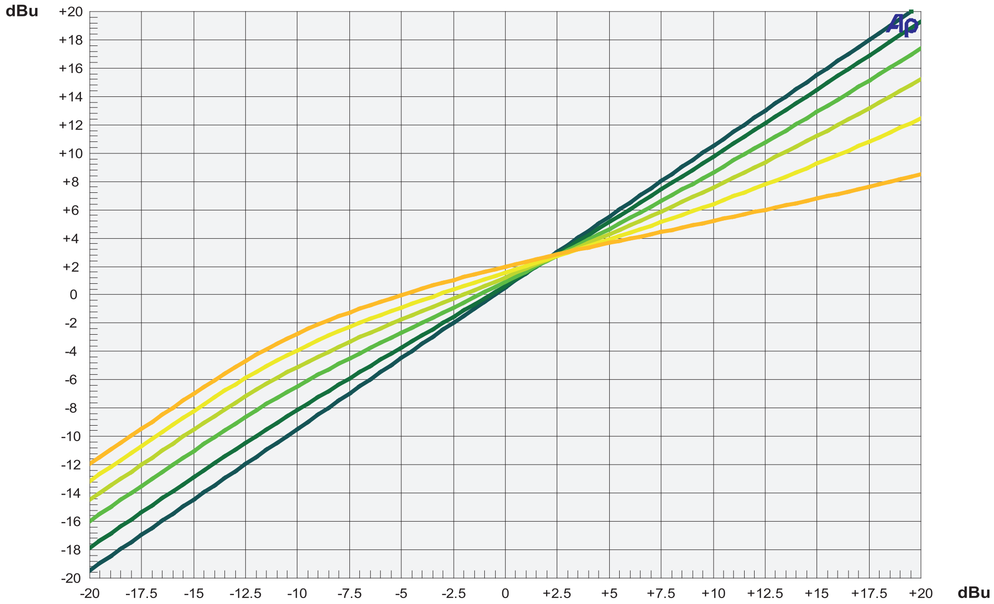
Here the controller was turned completely the other way round. In high pass position (HP) the approach of the sidechain filter is slightly different. At 100 Hz the compressor now hardly reacts to just this frequency and only the higher frequencies will have an increasing influence on the gain reduction. In practice the controller should therefore be set to the lowest frequency you just about want the compressor not to react to.




REFERENCE – SC Low Pass & SC High Pass

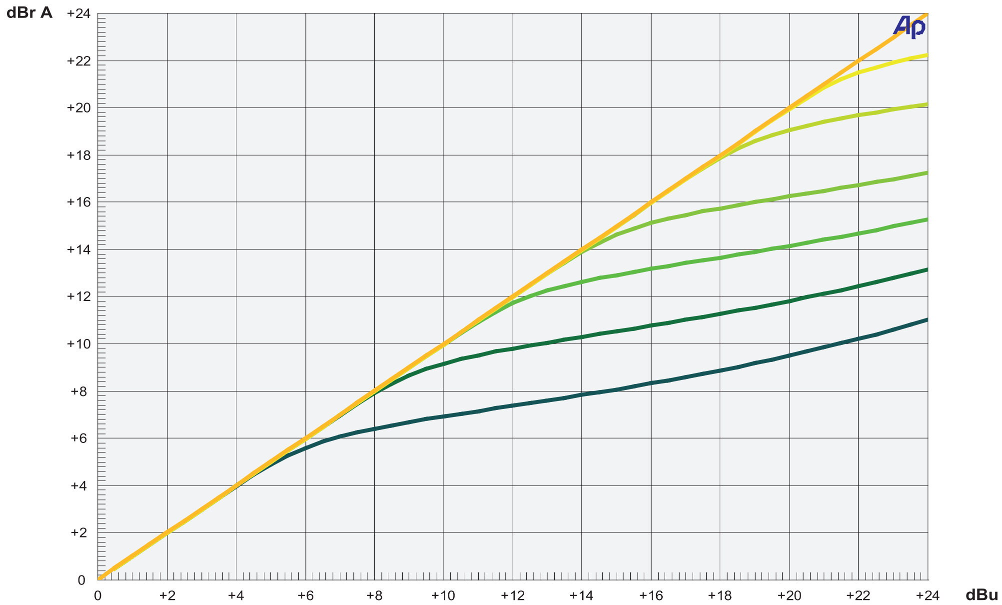
## Mix

This measurement diagram shows a fade from the original to the compressed signal (direct and compressed buttons active). The gain controller was set at 8 dB, which is also the reason for decreasing levels in the more clockwise areas of the controller range. In case that only the direct or alternatively only the compressed signal is switched in, the mix controller does not have any influence on the level.




## Soft Clip

These characteristic curves show the smooth processing of the Soft Clip limiter. Even though this figure employs huge amounts of level reduction to demonstrate the effect clearly, the limiter values should not exceed 6 dB as a general rule. After all, this function aims at rounding the peaks of a signal and not at reducing the complete level. Too heavy use of the Soft Clip limiter can cause audible distortion.



## Level Problems

### No Signal

This problem might be caused by balanced wiring. If the signal almost completely drops when the alpha compressor is activated, it is probable that a pin of the XLR connector at the input is not connected. As the input stages feature transformers, both pins have to be connected. The classic case for this problem would be a balanced XLR connection that is connected to an unbalanced output that only uses ground and pin 2. Connecting pin 3 to ground should solve this problem.

### Slight Level Jump

Under certain circumstances, pushing the active switch may cause a small change of level. If there is no noticeable difference between bypass and active when in stereo mode with activated direct buttons and mix controllers fully turned counter-clockwise, everything is OK. But if there is a small change of level somewhere between 0.5 - 1 dB, the input and output impedances of the devices preceding and succeeding the alpha compressor might be the reason. A high output impedance (approx. 1 kOhm) of the preceding device and a low input impedance (<4700 Ohm) of the succeeding unit are especially critical. If the compressor is in bypass, the preceding device will be loaded with the input impedance of the alpha compressor itself plus the input impedance of the succeeding unit. This cumulative load might result in a decrease of level. If now the alpha compressor is activated, the preceding device only "sees" the alpha compressor and no longer the following unit. Therefore the load decreases and the level rises as a consequence. Especially tube equipment tends to produce this phenomenon. Generally speaking, output impedances smaller than 150 Ohm are recommended for a trouble-free signal chain.

### Large Level Jump

The output stages of certain audio processors are designed in a way that the level will always stay the same – no matter if they are hooked up with a balanced or an unbalanced connection. If pin 3 is connected to ground, for example, the level at pin 2 will automatically become twice as loud as before. These kinds of output stages are usually unproblematic. But there are also stages that cannot compensate for that. In these cases the level at pin 2 stays as it is, even if pin 3 is connected to ground. If the alpha compressor is placed between a device with that kind of output stages and a device with unbalanced inputs and with pin 3 connected to ground, it is possible that the level jumps up by 6 dB when the compressor is activated. As a general rule, input stages with balanced connectors are always the best choice. In case these are not available, the first attempt to solve this problem should be to disconnect pin 3 at the XLR inputs of the alpha compressor and then connect it to ground. This generates an unbalanced signal that should not shift the level anymore.

## Technical Data

| | |
|---|---|
| **Frequency Response:** | <10 Hz – 200 kHz (-0.5 dB) |
| **THD+N @ +15 dBu, 20 Hz - 22 kHz:** | |
| Stereo Mode (Direct) | 0.0039 % |
| Stereo Mode (Compressed) | 0.009 % |
| M/S Mode (Direct) | 0.014 % |
| M/S Mode (Compressed) | 0.034 % |
| **Noise Floor, 20 Hz - 20 kHz (A-weighted):** | |
| Stereo Mode (Direct) | -95.8 dBu |
| Stereo Mode (Compressed) | -89.3 dBu |
| M/S Mode (Direct) | -95.6 dBu |
| M/S Mode (Compressed) | -92.3 dBu |
| **Dynamic Range, 20 Hz - 22 kHz:** | |
| Stereo Mode | 122 dB |
| M/S Mode | 118 dB |
| **Maximum Input Level:** | |
| Stereo Mode | +28 dBu |
| M/S Mode | +23 dBu |
| **Maximum Output Level:** | |
| Stereo Mode | +27 dBu |
| M/S Mode | +23 dBu |
| **Input Impedance:** | 10 kOhm |
| **Output Impedance:** | 68 Ohm |
| **Pin Assignment Input:** | 1. GND<br>2. Positive (transformer balanced)<br>3. Negative (transformer balanced) |
| **Pin Assignment Output:** | 1. GND<br>2. Positive<br>3. With 68 Ohm to GND |
| **Power Consumption:** | 100 W |
| **Fuse Type:** | 115 VAC 3 A Slo-Blo/230 VAC 1.5 A Slo-Blo |
| **Dimensions (W x H x D):** | 483 mm x 133 mm (3 U) x 405 mm<br>19“ x 5.3“ (3 U) x 16“ |
| **Weight:** | 16.0 kg/35.3 lb |

## Warranty

**Conditions and limitations**

The alpha compressor is covered by a limited warranty for a period of 5 years against defects in parts and labor from the date of purchase. Natural wear is not covered by this warranty. elysia will remedy problems caused by material or workmanship either by repair or replacement to restore the product to full performance without charge for parts and labor. Repairs or replacements will not extend the warranty period.

The warranty is given to the original purchaser only and is not transferable. elysia will only give warranty on products purchased through authorized elysia dealers. The warranty will only be valid in the country of the original purchase unless otherwise pre-authorized by elysia.

All warranties become void when the product has been damaged by misuse, accident, neglect, modification, tampering or unauthorized alteration by anyone other than elysia authorized service personnel.

The warrantor assumes no liability for property damage or any other incidental or consequential damage whatsoever which may result from failure of this product. Any and all warrantees of merchantability and fitness implied by law are limited to the duration of the expressed warranty.

This warranty gives you specific legal rights and you may also have other rights which vary from state to state. Some of the above limitations may not apply to you.

**Registration and return**

To confirm your warranty please register your elysia product shortly after purchase. The easiest and fastest way is to use the online form you will find in the service section of our website www.elysia.com. If you cannot or do not want to use this option, please contact us via the address you will find at the end of this warranty statement. You will be provided with registration documents suitable for fax or mail use free of charge.

In case you notice any defect, please contact elysia for technical support directly. You can find the correspondent contact data at the end of this warranty statement. You will receive a return authorization which enables you to send your product to the elysia factory where it will be repaired and then sent back to you.

**Packaging and shipping**

All returns to the factory must be in the original packaging, accompanied by the return authorization, and must be shipped via insured freight at the customer's own expense. A new original packaging can be ordered from elysia. The customer will be charged for new factory original packaging if he fails to ship the product in the original factory packaging.

In case that a product must be returned to the factory from a country outside Germany, the customer shall adhere to specific shipping, customs, and commercial invoicing instructions given with the return authorization as elysia will not be responsible for transportation costs or customs fees related to any importation or re-exportation charges whatsoever.

After repair, the product will then be returned to the customer via prepaid, insured freight, method and carrier to be determined by elysia. elysia will not pay for express or overnight freight service or pay for shipments to locations outside Germany. All damages caused by transport are not covered by this warranty.

**Contact data**

For technical support and return authorizations, please contact:

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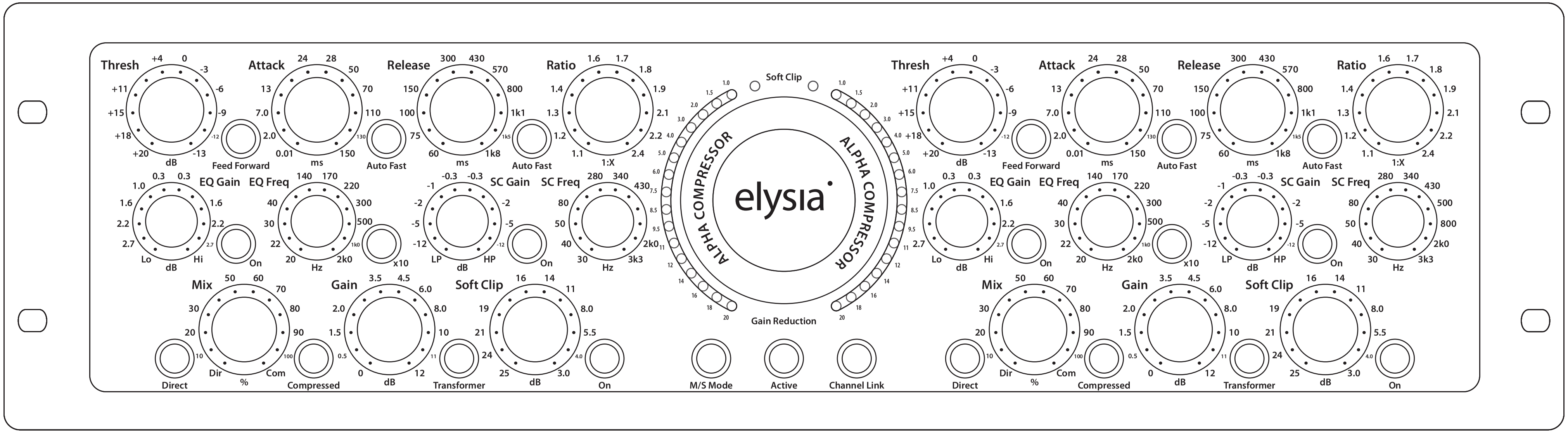
P.O. Box 20 10
D-41307 Nettetal
Germany
Tel: +49 (0) 21 57 / 12 60 40
Fax: +49 (0) 21 57 / 12 63 12
service@elysia.com




Project . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . .

Song . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . .

Engineer . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . .

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Left Channel Track . . . . . . . . . . . . . . . . . . . . . . . . . .

Right Channel Track . . . . . . . . . . . . . . . . . . . . . . . . .

Notes . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . .

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„Let there be sound“

And there was sound.

(AC/DC)