# Evaluation Therapy

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## Tomo Audiolabs LISA Mastering EQ – a final report


Exactly one year before, in the March 2009 version, I presented you a very extravagant Mastering and Mixing /EQ, which interpreted anew, the principle of a frequency-dependent dynamic processing in a very special way. Unlike the dynamic equalizer or multi-band compressors, the adjustment to a particular position is carried out directly in the filter, and not behind it. In this way, the compressor and the expander do not react on purely frequency-selective mode, but they affect directly the amplitude and the performance of the peak filter, or the trans-conductance and lug frequency for inclination filters of outside hinges. The dynamic processor, located in each of the six bands of the two-channelled device, works according to the feedback principle with an optical adjustment element, and therefore it takes its adjusting signal from the filter output which results in the typical stabilizing process of reversely regulating compressors. Another special feature of this device is the controllability of the original signal, which for usual filter designs is „passed-through“ within the additive or subtractive process of a filter. It would not be that correct to designate this function as „parallel compression“, but it is the best way to illustrate this principle . The pure filter signal, dynamically adjusted or unadjusted, is carried out in a separate way, which in an extreme case can be intercepted / used completely without the original signal component. In this way, while on one hand the complex regulatory processes can be controlled as in the „Lists“ function like for e.g. in a „De-Esser control“, on the other hand exotic signal structures can also be generated, which through the continuous mixing of the original signal become more and more evident. Why do I tell you this story? Last year, the developer Helmut Butz has worked intensively on detailed improvements and technical optimizations, which I now want to present to you as a final version of the LISA EQ in the frame of a second, final test.

Since I do not want to start from the scratch, I will limit myself to some core statements for a better understanding of the relations, and recommend you to read the initial test of the March 2009 version. Subscribers of the Studio Magazins can find this test also in the reader account domain on our website - if they haven't done it already - by entering their six-digit customer number as user name and password. For security reasons, after the first log in, I recommend you create your own password in the personal data domain.

## Short Profile

Besides the features that have already been sketched out in the introduction, LISA can be optionally applied in two-channel, interconnected and M/S operations. This means that the user has a high-quality, sensationally sounding six-band stereo-and M/S-EQ for production and/or mastering, which additionally makes separate dynamic functions for each band available in a manner that has been described already. The outside bands are designed as peak filter with compression function (boosting only), and the four parametric bands with compressor and expander functions (boosting and depression). Additionally, in the input domain there is a low pass filter that can be switched on. Without exception, all controls are designed as a switch, thus allowing a precise repeatability of settings, or a quick comparison of both channels in stereo mode. **Figure** illustrates the impact of the dynamic working Expand function on an increase in filter set at 1 kHz: frequency: The higher the input level, the stronger and simultaneously steeper the boost produced by the filter. In order to understand the diagramm, it is important to know that all measurements were done with identical device settings. The input level induced to EQ by measuring systems presented some variations. The general principle is that, increases in EQ are reduced by the compressor according to its settings and the expander is strengthened. During depressions in the EQ, the compressor reduces its depression and the expander boosts it. More detailed information can be found in the initial test.

## What's new?

To know this, I conducted a detailed interview with the developer Helmut Butz, with whom I could discuss on this occasion my latest listening experiences. The desire for improvement and optimization was built partially at pure technical level, through the Tomo-Team's own quality standards, but also through the user feedback and, I must say in all modesty, by criticisms or suggestions on our first tests. A basic change was carried out for the electricity supply, which is now taken over completely by an external power supply. In the same context, a technical change was made in the input circuit. The resistance network for the input gain is now located behind the input transformer. This asymmetry of the values for the attenuation significantly improved and linearized the frequency response, which had some low frequencies (see measuring Instruments). A critical point represents the interference spectrum observed by us, which now looks considerably better through the usage of another ring core (toroidal) transformer and an optimized ground loop. Although the measurements provided for the overload at 1 kHz have showed almost no other results, the characteristics of the transformer, to which the input signal is then fed directly, show certain ‚problems' at very low frequencies. There the overload is reduced to a value of approximately =15 dB, which you can certainly use as a design tool, if you want to „equip" this domain with some distortion factors.


The filter design was revised again in different points. The properties of the filters have been optimized for quality and distribution, especially in the high peak filters, the frequencies were slightly decreased. While switching between frequencies, the amplitude values were homogenized and he in-and out oscillation procedure was optimized , which contributes to a softer and more noble ‚sound impression. Moreover, the lug (approach) frequency of the peak filter is not shifted that strongly due to the compressor activities, and hence the modulati-

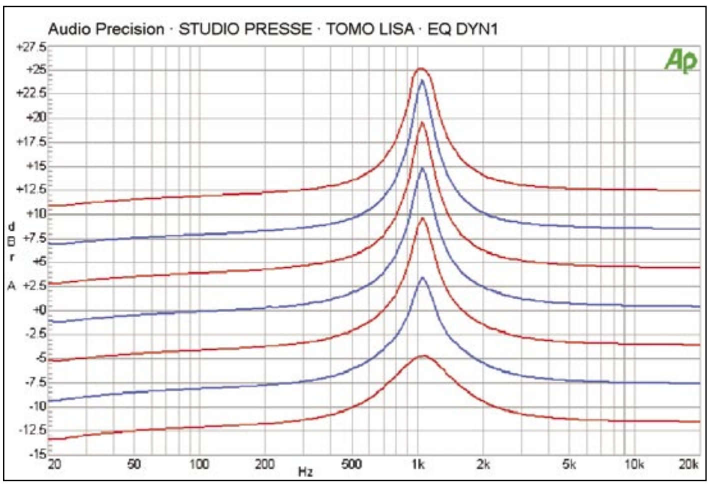


*Diagram of the expander function*

*External power supply for LISA...*

*plus Siemens connector and dim control*

on effects observed by me are no longer practical. Low-Mid- und Hi-Mid-bands are equipped with different condensers. This is why Hi-Mid sounds softer and more pleasant, while Low-Mid sounds a bit more „aggressive‘.

During stereo interconnection the stereo panorama (image) has become considerably more stable. To optimize this, the control voltages were coupled with each other in a different place. The optocoupler carries out its work in the Negative feedback branch of the filter. This was initially used in combination with a series resistance, and thus covered, only a portion of the control resistance. This series resistance was removed, so that a new optocoupler takes over the whole range of control. Since this is driven not so strongly , the dynamic bandwidth has increased. This results in a total fine tuning possibility and a less branchially designed control behaviour in the domain limits. This way, one can move more sensibly in extreme areas. Practically it proved to be advantageous to set both operating point values equal in the adjacent filter bands. In principle it is noticed that the channel with the higher control drags the other channel along with it. If the operating point of a channel is zero or is switched off, then the other channel will be set to zero as well.
On the part of ergonomics, the scale label was optimized and is now much easier to read . The red light strip that separates the two channels is indistinguishable in appearance, but can be made dimmer with a knob on the back, which increases the readability of the scales. Besides, the entire user interface is backlit by other lighting elements.
The activity of the dynamics processor is now shown by a separate LED, which was previously integrated in the power button for the dynamics from which it was not clear enough for the user, whether the dynamic action was active or not. It will also give a special mastering version with reduced amplitude values, which will allow higher resolution level settings.

### Measurement results

Our measurements on the actual device first showed a correct response in the depth range (Figure 1), which with a fall of approximately -1.6dB at 20 Hz was not entirely homogeneous. The level and phase frequency access are now ruler flat in almost the whole hearing domain, however beyond 20 kHz a soft level increase appears, and with about +16 dBr it reached its maximum at around 95 kHz. TThe distortion factor moves up to the input and output levels of around +22 dBu is excellent below 0.005 percent and rises to the maximum level of +28 dBu with a soft course to around 0.15 percent - only beyond this, it‘s really ‚uncomfortable‘. With the internal gain at zero, all the filters and regulators reached the Unity Gain almost perfectly with +0.05 dB. The output noise in neutral position is - 87.2 dBu RMS effectively une-

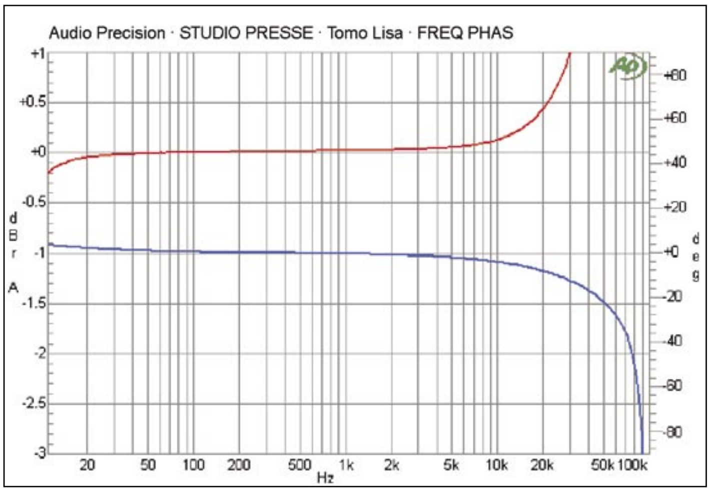


*Diagram 1: Level and Phase Frequency Access, all filters in neutral position*

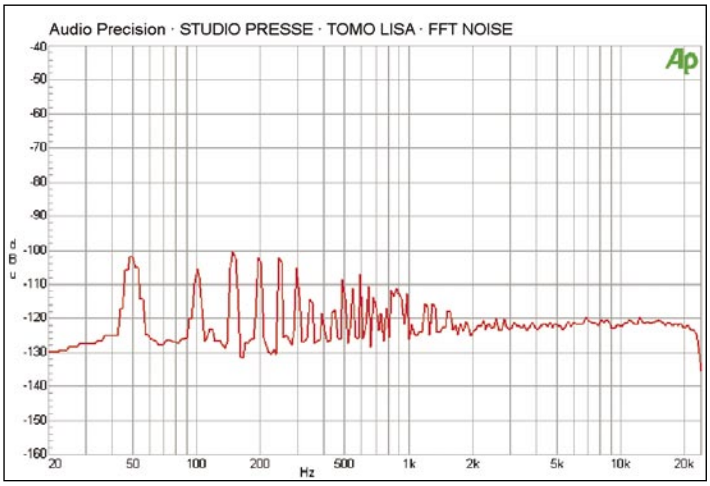


*Diagram 2A: Measuring of the FFT Noise Spectrum in March 2009*

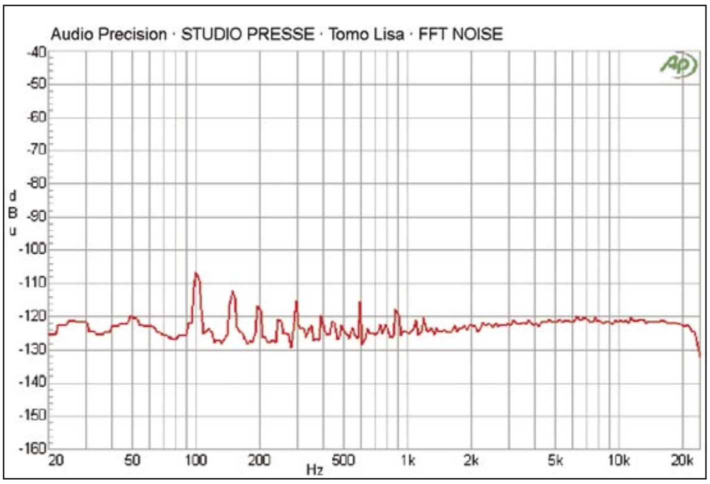


*Diagram 2: FFT-Noise Spectrum, all filters in neutral position*

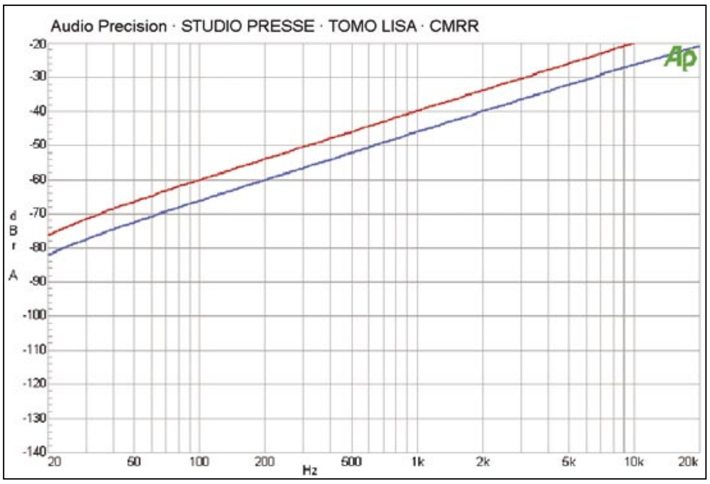


*Diagram 3A: Measurements of the Unbalance Attenunation of Inputs in March 2009*

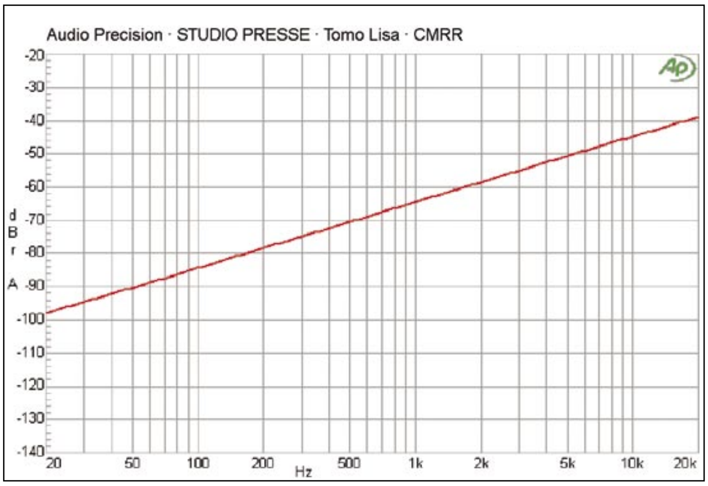


*Diagram 3: Unbalance Attenuation of Inputs*

valuated (22 Hz up to 22 kHz), which is marginally worse than the first test device: for a dynamic domain of 115 dB, this is pretty painful. The quasi-peak measuring with CCIR filter resulted in -75.9 dBu. Diagram 2 shows the FFT noise spectrum; at 100 Hz the interference (ripple) components reach a maximum of only -108 Hz, an increase of about 6 dB lower than those for the first test device. The rejection of inputs sown in Figure 3 are significantly better; at 1 kHz, -65 dBr are reached. This means an improvement by 20 to 25 dB.

## Practice and Hearing Impressions

In order to check the current sound features again, I spent one hour in our studio. I had some fun, thanks to the listening sessions that were much longer than planned. The device is now in its final circuit state. For the series, some mechanical improvements only shall be made, but the device functionality and its technical status may not be touched under any circumstances. For the test I had both finished products as well as some unprocessed material. By this, „docile' or ‚better managed ‚control behaviour of the dynamics processors, the dynamic spectral editing has become much faster and handier. The filling of spectral dips is thus a real pleasure, as the dynamic control of each strip is active over a very wide area, with full control from virtually a zero boost (increase) to a maximum retraction of the boost through the compressor. Especially in combinations with an uneven depth domain a perfect homogenized spectrum can be obtained. The Hi-Mid filter allows for „the meanest" frequencies in the domain of 3 or 4 kHz, also without dynamic control. The treble tilt filter now sounds much softer and gives a mixture of open, silky shine. In order to refer once to the headline of this reference paper: The real magic with this device is the spectral movement, which offers each a blend of attractiveness and assertiveness, which I could not obtain with any other device. The mixture of static and dynamic bands provides an extremely flexible adjustment, which in most cases, if you are not known as a loudness fanatic, makes the use of an additional broadband compressor or loudness-servant superfluous. However, this works best with well mixed material. The use of the M / S mode gives the device an added dimension, thanks to the concept that leads to results which could not be produced so far, especially not on such a simple and direct way. Therefore, we are sure that after long but reasonably used development time, Tomo Audiolabs has now reached the destination of its ideas.

## Conclusion

Thanks a whole package of improvements, changes and optimizations LISA has now been elevated to a much higher quality level, as this was already evident even in pre-production model, both in terms of technical data, and the ergonomics and functionality. In spite of its active design, EQ helps the unique signature of a passive design, but may also strongly wish to grip the audience. The control behaviour of the dynamic processor feels considerably better in practical use, and the ergonomics has undergone major improvements that make the device easier to control. What hasn't changed is the complexity of the functions and the resulting demand for very experienced users, who can realize the value of this device and are able to appreciate it. Neither was there any change of price, which a serious and prospective customer may continue to assume in the five figure range. Such a massive and complex phenomenon like LISA is, however, worth each Euro, considering material and development expenses. what was an year ago looked at as a possible future device, has now become reality. I'm still convinced that with its unique features, LISA has earned the highest attention among professional users. This very special EQ, or better said Signal processor, is definitely into the „Hall of Fame" of professional audio-technology. ■