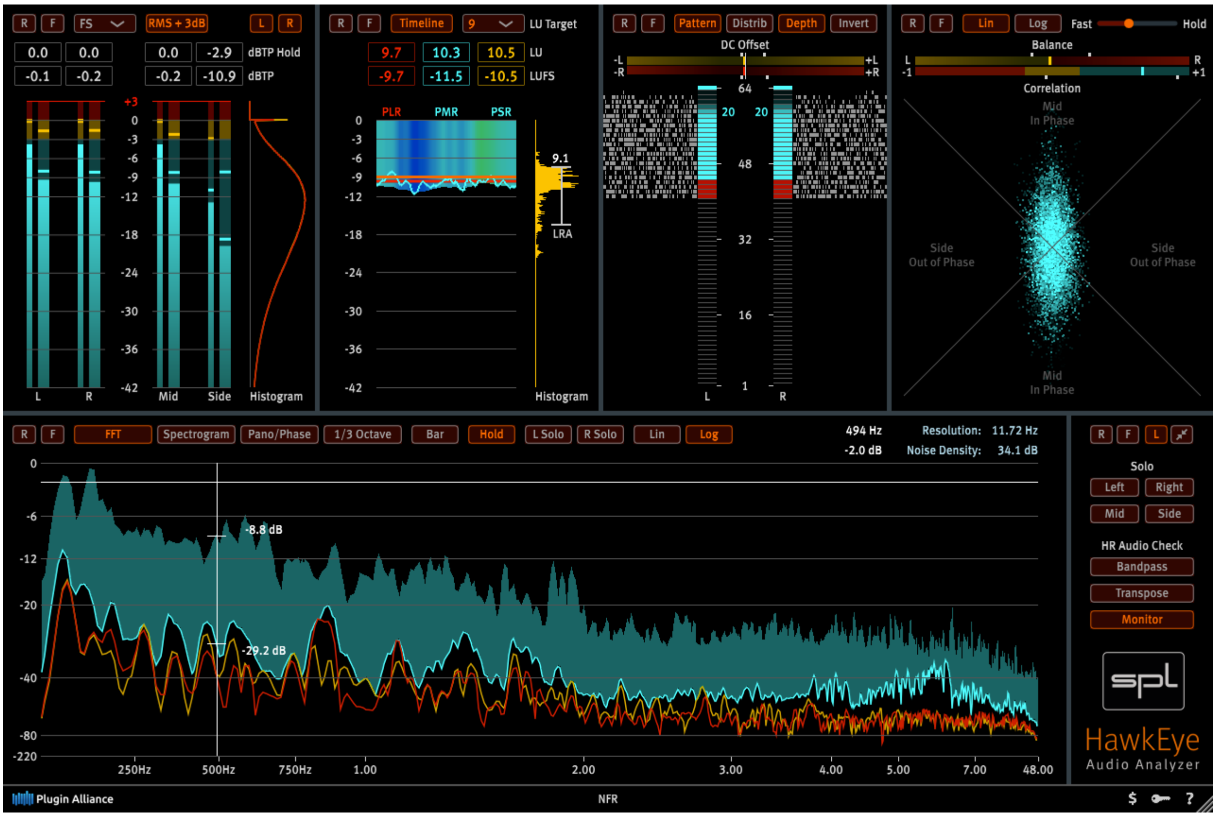
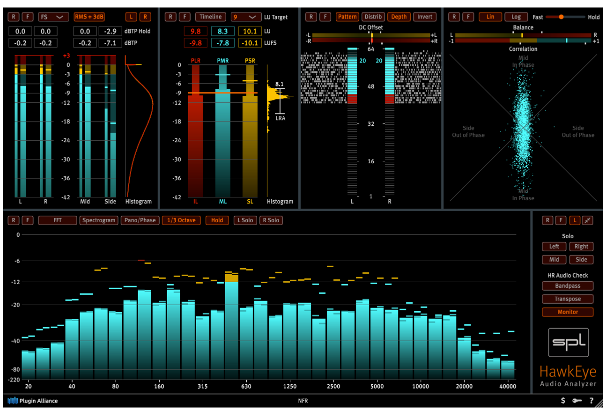
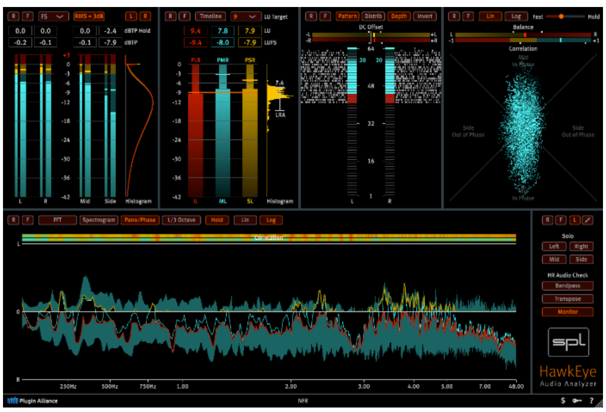
HawkEye

Audio Analyzer

Manual







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Introduction

HawkEye comprises the most versatile audio measuring instruments in one toolset. It has been designed and developed with highest precision in mind to set a new standard in audio analyzing.

Unique features like bit-depth analysis and high frequency monitoring offer a new window into the world of high resolution audio recordings.

The user experience follows aviation industry guidelines for designing cockpit instruments. HawkEye presents all information uncluttered, minimizing eyestrain over extended periods of use.

Feature Overview

- Supported sample rates of up to 768kHz to cover todays high resolution audio content
- Full double precision 64 bit processing
- Five measuring instruments:
  - Level Meter
    - True Peak meter for L, R, Mid and Side
    - RMS metering for L, R, Mid and Side
    - TPL histogram
    - dB full scale as well as the Katz-Scales
    - dBTP hold and falling numeric displays
  - Loudness Meter
    - PLR – Peak to Loudness Ratio
    - PMR – Peak to Momentary loudness Ratio
    - PSR – Peak to Short-term loudness Ratio
    - IL – Integrated Loudness
    - ML – Momentary Loudness
    - SL – Short-term Loudness
    - Timeline display -> PLR, PMR, and PSR (color coded) over time
    - LU Target selection
    - Loudness Range Histogram

- Bit Monitor
    - 64 bit display
    - Bit depth check - Determine real bit-depth
    - Bit pattern timeline display
    - Bit distribution timeline display
    - Bit scale inversion
    - Direct Current (DC) offset meter (-60dB to -40dB)
- Vector Scope
    - Goniometer
    - Linear and logarithmic display
    - Luminescence slider
    - Balance meter
    - Correlation meter
- Spectrum Analyzer
    - includes Spectrogram, Pano/Phase and 1/3 Octave view
    - Spectrum (FFT) with 4096 bands, applying a 7th order Blackman-Harris-Window to achieve a great frequency and amplitude precision
    - Spectrogram (spectrum over time)
    - Panorama- and Phase-Display with a resolution of 4096 frequency bands
    - 1/3 Octave Analyzer
    - Linear and mouse draggable logarithmic frequency scale
    - Mouse draggable dB scale to change the displayed amplitude ranges
    - L/R Solo

- All 5 measuring instruments can be undocked from the main window to freely resize and position them over multiple screens.
- High Resolution Audio Check
    - Bandpass selection to isolate especially high frequency content.
    - Shift the selected frequencies into the audible frequency range for acoustic monitoring.

General Functions

Undocking & Resizing

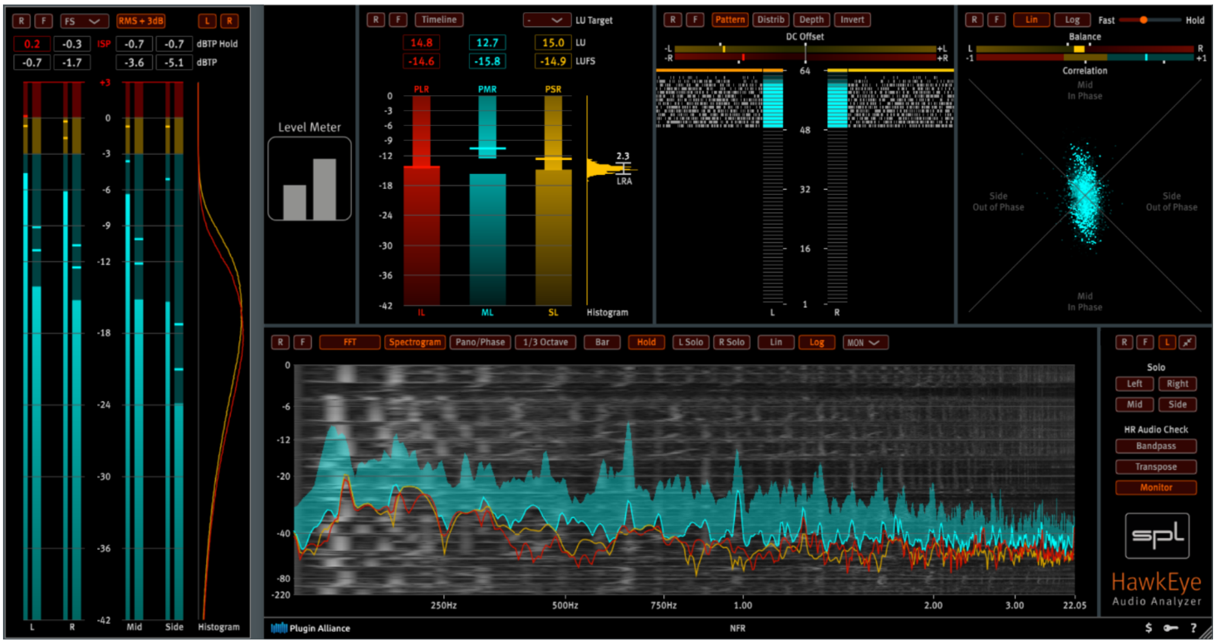
HawkEye is designed to show all measurement instruments within a single window, yet this is only one mode of operation.

To be as flexible as possible any instrument can be undocked in its own window by a simple double-click.

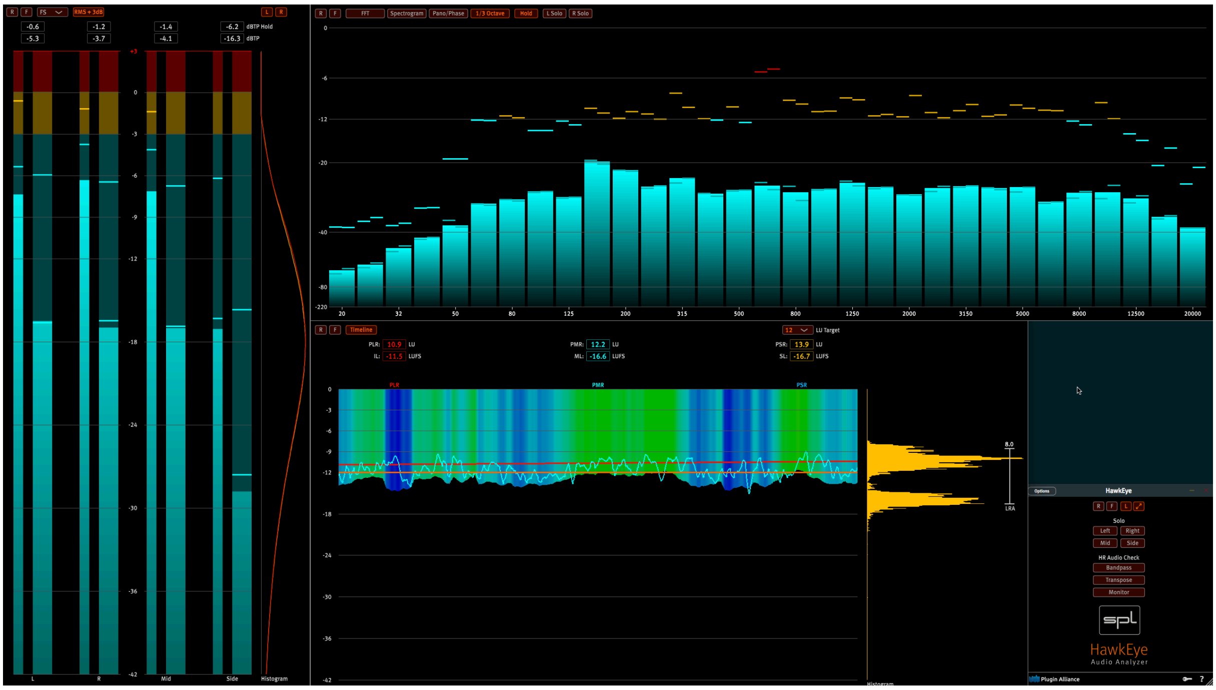
Furthermore, each instrument window is freely resizable and can be positioned on multiple screens.

A double-click on the undocked instrument window or its placeholder docks it back into the main window, whereas a single click on a placeholder brings the associated instrument window back into the foreground.

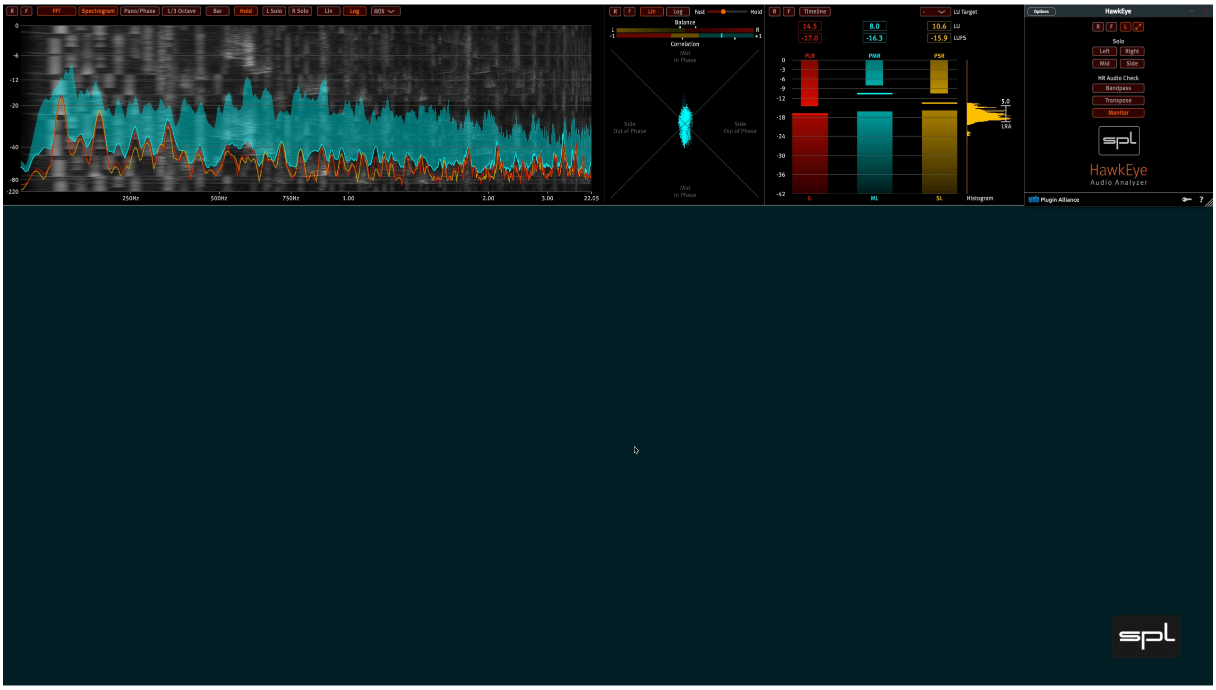
All configurations are storable as presets to be re-called at any time.



[1] Undocked Level Meter



[2] Minimized Control Panel and undocked Level Meter, 1/3 Octave Analyzer and extended Dynamic Timeline windows



[3] Minimized Control Panel and undocked FFT with Spectrogram, Vector Scope and Loudness Meter windows positioned at the top of the screen

Reset & Freeze

All instruments provide a Reset [R] and a Freeze [F] function.

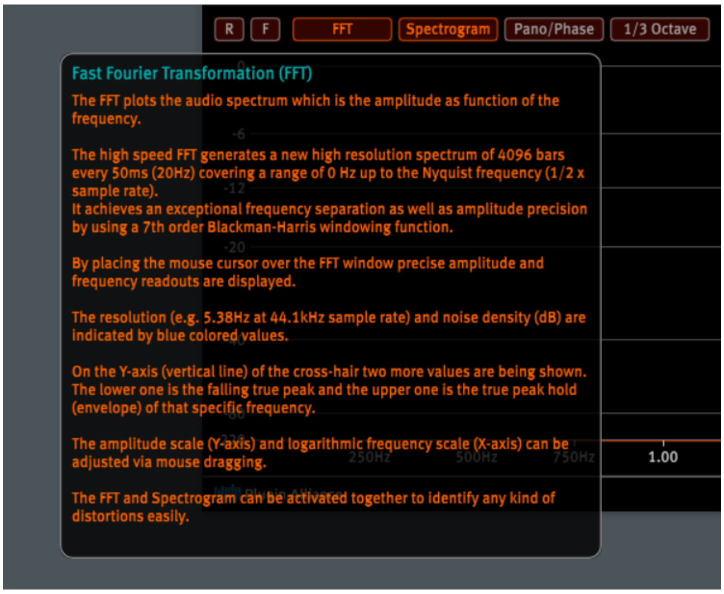
[4] Reset & Freeze

Reset clears all values of the instrument and restarts the measurements.
Freeze freezes the display while the measurements continue in the background.

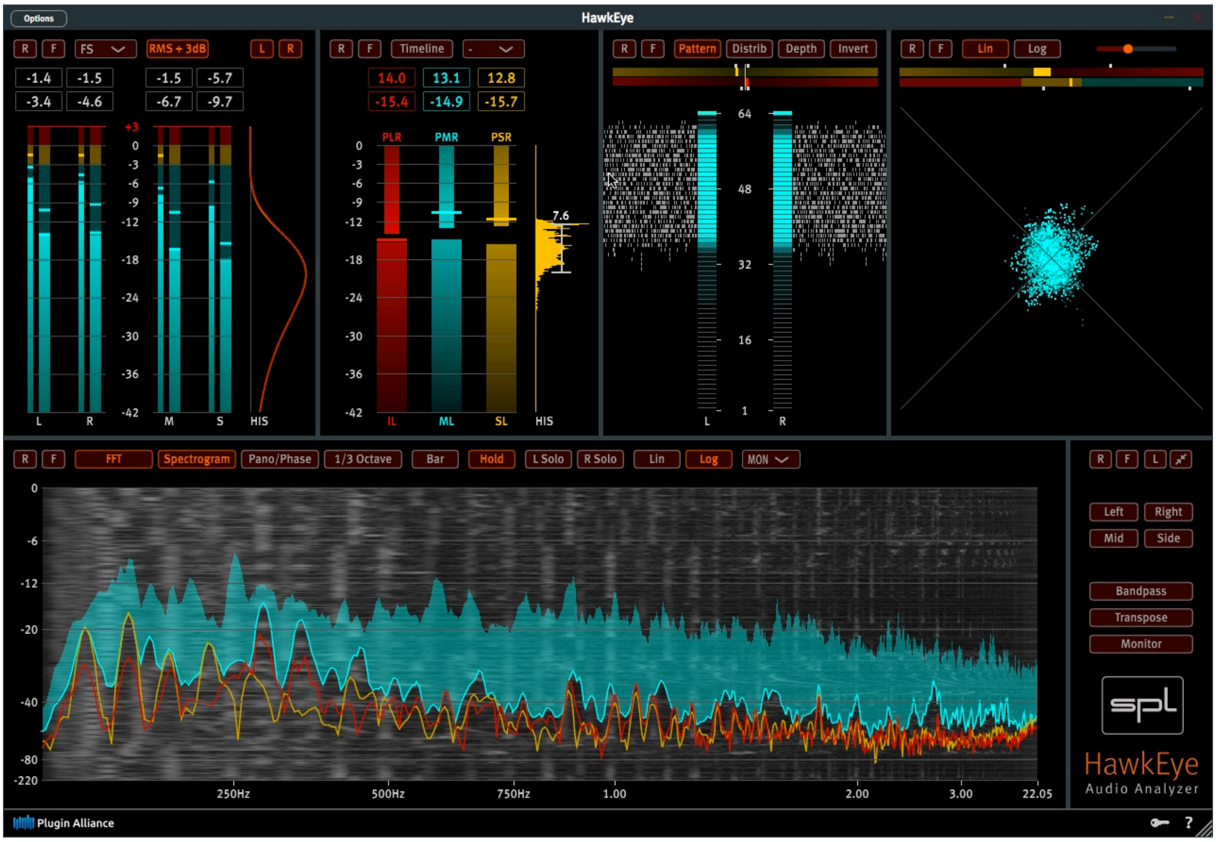
Tooltips and Labels

Buttons, drop down menus as well as numeric displays show extensive tooltips when the mouse rests on them for a short while.

This function as well as additional labels can be switched off to further declutter the instrument windows.



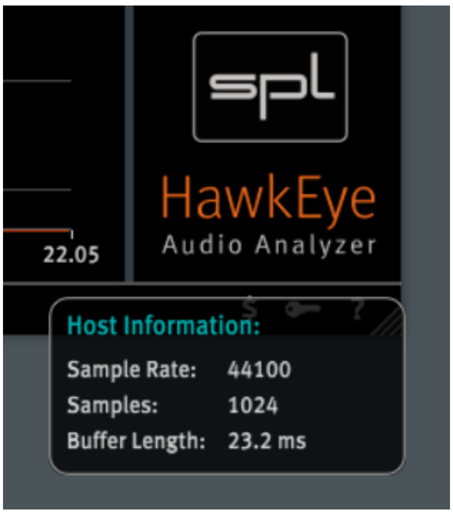
[5] Tooltip example



[6] Stand-alone version with Labels deactivated for the most uncluttered appearance

Host Information

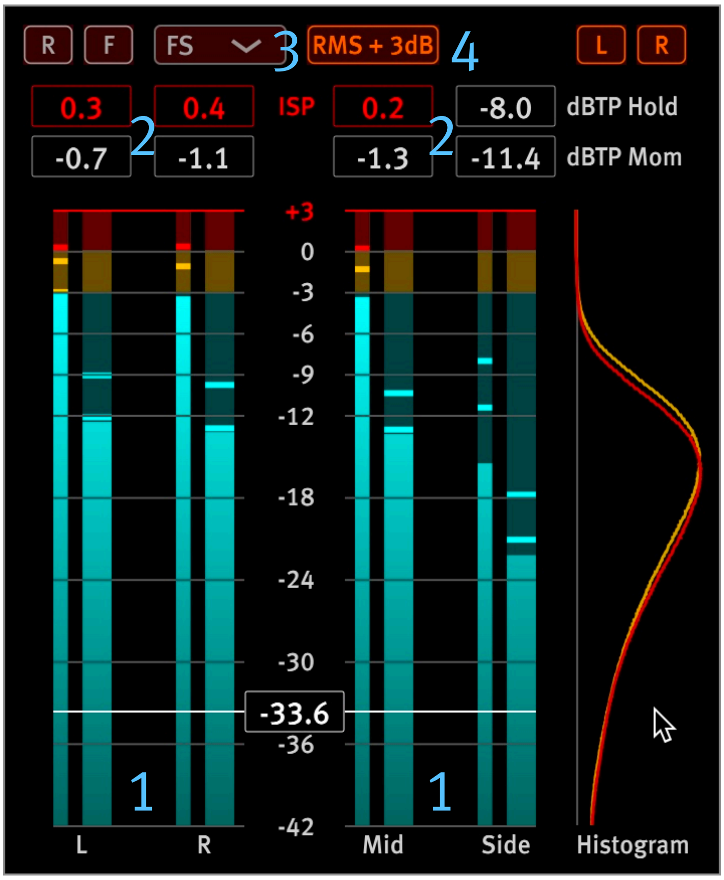
By hovering the mouse pointer over the lower part of the SPL HawkEye logo the following information is displayed within a popup-window:



- Sample Rate
- Buffer length in samples
- Buffer length in time

[7] Host Information

Level Meter



[8] Level Meter window

1 True Peak & RMS level meters

The slender meters show the TPL (True Peak Levels) and the wider meters display the RMS (Root Mean Square) levels.

Beside left and right the Level Meter displays the mid and side signals as well which are useful to get a better idea about issues within a stereo signal.

The mid [M = sum = (L+R)/2] contains the in phase mono part of the stereo signal. The side [S = difference = (L–R)/2] contains the out of phase part of the stereo signal.

A precise readout is provided when moving the mouse cursor over the instrument window. A white horizontal line appears with the value in a box.

2 True Peak Hold & True Peak Falling

The upper row of value boxes contain the maximum true peak hold levels of the respective channel. Those values are also displayed by the top horizontal lines in the meter bars.

The lower row of value boxes show the falling true peak levels. These values are also displayed by the falling horizontal lines in the meter bars.

The oversampled TPL meter (True Peak Level) can detect Inter Sample Peaks (ISPs) which are not visible on a standard digital meter.

ISPs emerge as soon as a recorded signal is not limited by an oversampled limiter and hits the 0dB threshold.

ISPs are an issue when an audio signal is converted back into the analog domain where they can cause overs of up to 3dB resulting in distortion within the analog part of the DAC.

An appearance of an Inter Sample Peak is indicated by a red ISP symbol in the upper row of value boxes. Both the upper and lower value box will turn red in the event of ISP detection.

3 Scale Selection

The Level Meter supports four scale types:

1. FS (Full Scale) in dBFS: -42 to +3dBFS (includes 3dB for Inter Sample Peaks)
2. K-12: -30 to +12dB (+ 3dB for ISPs)
3. K-14: -28 to +14dB (+ 3dB for ISPs)
4. K-20: -22 to +20dB (+ 3dB for ISPs)

The K-Scales developed by Bob Katz propose an integrated system of metering and monitoring that encourages more consistent levelling practices in film, broadcast and music productions.

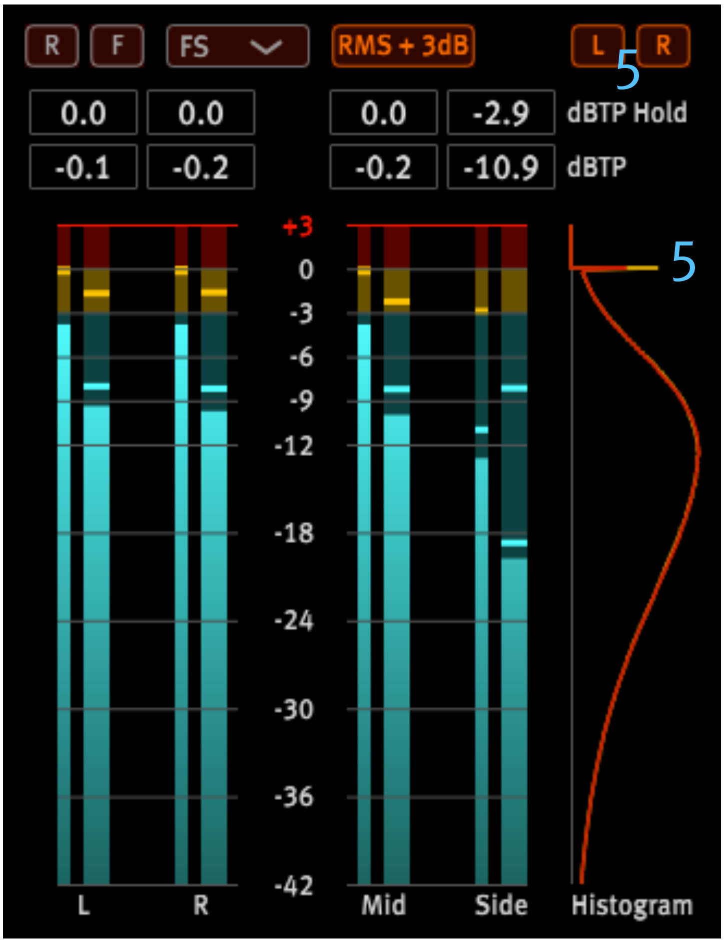
Click on the arrow to open the pull down menu and select your favorite scale.

4 RMS +3dB

Normally the RMS value of a sine tone is 3dB lower than the peak value. With RMS +3dB engaged the metering is EBU compliant, reading for example -20dB RMS as well as -20 LUFS when fed a -20dBFS stereo 1kHz sine tone.

Furthermore, when calibrating an audio system by using a sine tone it is most handy when the peak value is identical to the RMS value by activating RMS +3dB.

5 True Peak Histogram

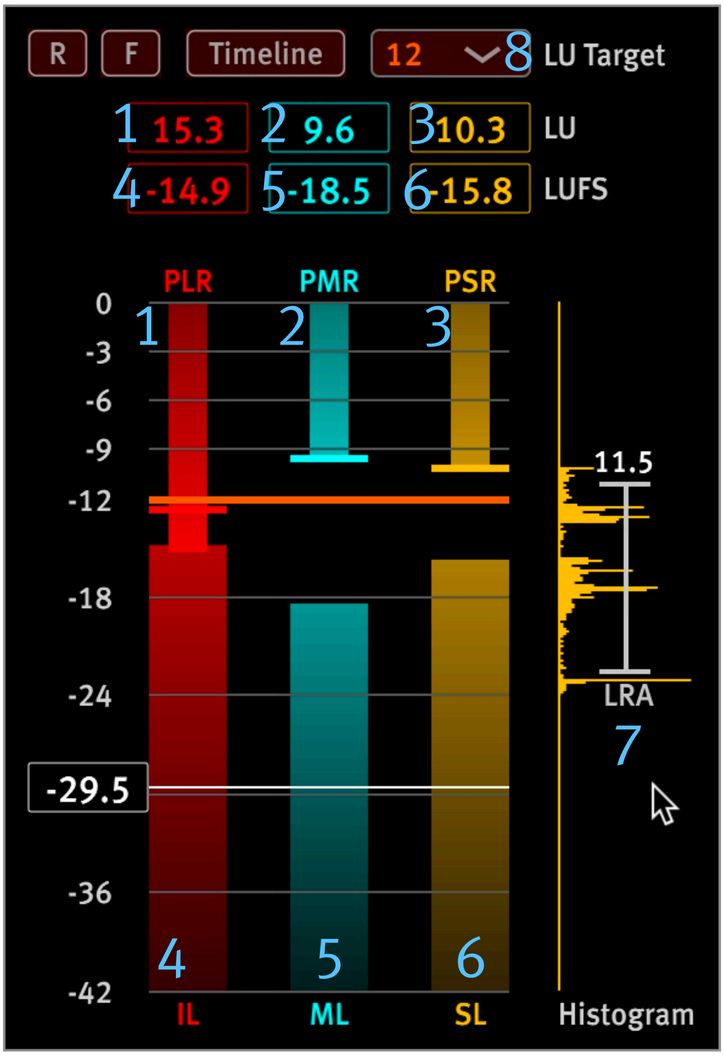


[9] Heavily limited audio track (see peak spike at 0dB)

The histogram displays the True Peak distribution of the left (yellow) and the right (red) channel. It allows to check whether the usage of compression or limiting stays within acceptable limits. A horizontal peak spike before 0dBFS reveals the extent of compression/limiting applied to the audio signal, whereas a gaussian distribution indicates a clean signal.

The L & R buttons switch the respective channel on and off in the histogram view.

Loudness Meter



[10] Loudness Meter window

1 PLR – Peak to Loudness Ratio

PLR is the ratio of maximum True Peak Level (TPL) to Integrated Loudness (IL). It is a long-term measurement in Loudness Units (1LU = 1dB) and is often used for loudness management by streaming platforms to ”normalize“ the playback level.

The PLR bar deflects downwards to make use of the same scale as the Integrated Loudness. The associated red value box located to the left in the upper row provides the positive LU value.

2 PMR – Peak to Momentary Ratio

PMR is the ratio of True Peak Level (TPL) to Momentary Loudness (ML) measured in Loudness Units (1LU = 1dB) over a 400ms window.

The PMR bar deflects downwards and its positive LU value is displayed in the petrol colored value box in the center of the upper row.

3 PSR – Peak to Short-term loudness Ratio

PSR is the ratio between the True Peak Level (TPL) and the Short-term Loudness (SL) measured in Loudness Units (1LU = 1dB) over a 3 second window.

This value is key to mix a track against a set target. By selecting a PSR target and activating the Timeline (9), it is possible to master towards a defined dynamic range.

The PSR bar deflects downwards and its value is displayed in the yellow value box to the right in the upper row.

4 IL – Integrated Loudness

IL represents Loudness Units relative to Full Scale (LUFS) over the entire time of measurement as defined by the EBU R128 recommendation.

The IL bar deflects upwards and its value is displayed in the red value box on the left in the lower row. The red horizontal bar holds the maximum IL value.

5 ML – Momentary Loudness

ML represents Loudness Units relative to Full Scale (LUFS) over a 400ms window of measurement as defined by the EBU R128 recommendation.

The ML bar deflects upwards and its value is displayed in the petrol value box in the center of the lower row. A petrol horizontal bar holds the maximum ML value.

6 SL – Short-term Loudness

SL represents Loudness Units relative to Full Scale (LUFS) over a 3s window of measurement as defined by the EBU R128 recommendation.

The SL bar deflects upwards and its value is displayed in the yellow value box on the right in the lower row. A yellow horizontal bar holds the maximum SL value.

7 LRA – Loudness

The yellow colored histogram displays the distribution of SL (Short-term Loudness) and indicates the LRA (Loudness RAnge) which is a measure of the variation of loudness on a macroscopic timescale.

The numerical LRA value is displayed above the vertical measurement bar.

8 LU Target

The LU Target refers to PSR (Peak to Short-term Ratio). It is a helpful tool when mixing against a loudness target.

The selected target loudness is displayed by a horizontal orange line across the loudness bars. By clicking on the drop down menu the LU target value can be changed in 1dB steps from -3dB to -23dB.

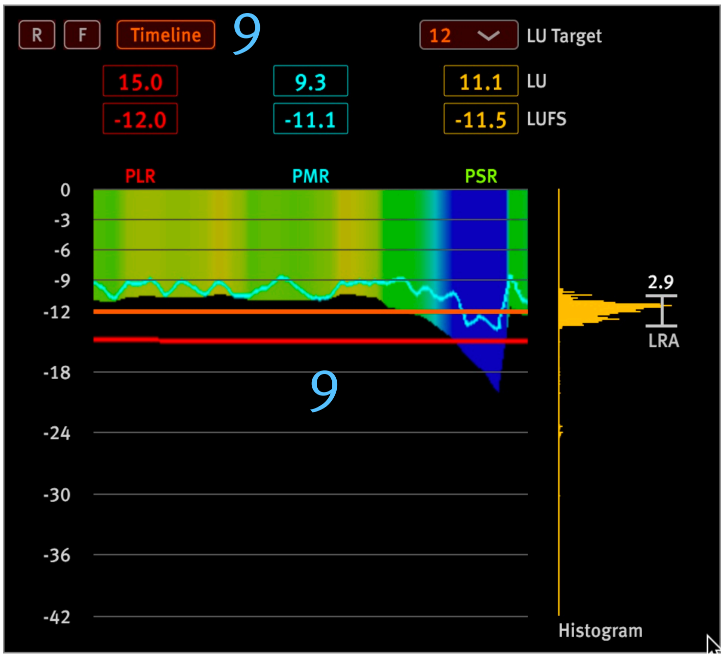
9 Dynamic Timeline

Clicking on Timeline activates the Dynamic Timeline display. It shows PLR, PMR and PSR over a certain amount of time. It is possible to cover a larger time frame by just undocking and resizing the instrument horizontally.

The PLR is shown as a red line and it is usually a pretty even line as this value changes slowly.

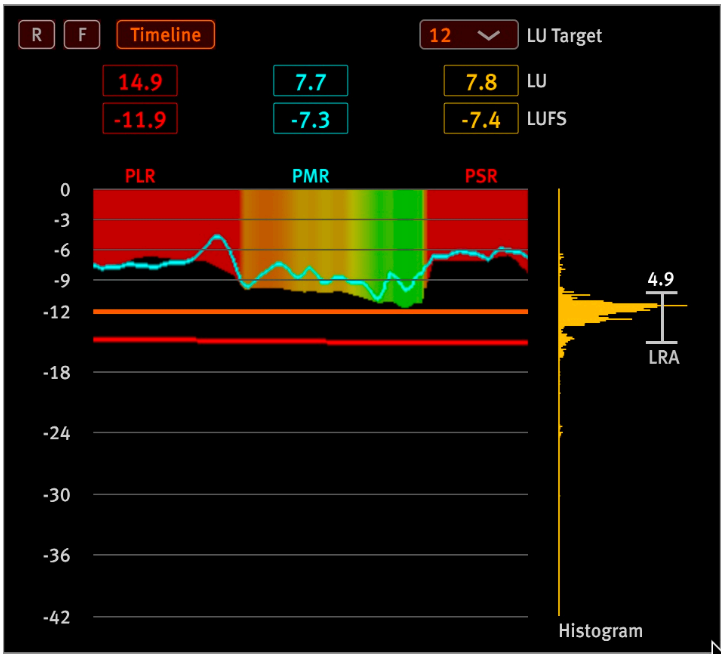
PMR is displayed by the petrol line. This is the most rapid measurement with just a 400ms window.

PSR is shown as a yellow field deflecting downwards. It turns into a color coded display when a LU Target (8) is defined.



[11] Dynamic Timeline window

The color coded PSR is the most important display when mixing against a loudness target. Blue indicates areas where the mix is cold and PSR is higher than the target allowing for more compression or limiting. The color chart gets green, when the PSR is on target and yellow when it is close to the target.

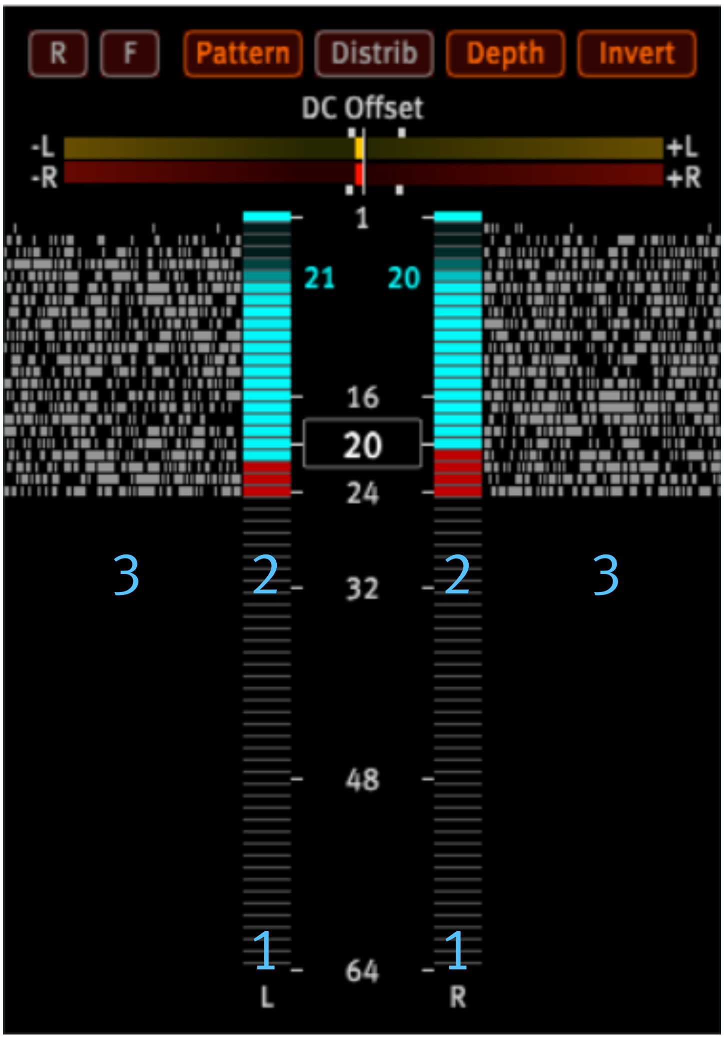


[12] Example shows that track's PSR is far off target

In the example above there are orange areas indicating that the mastered audio track drifts away too much from the target. When it is red the mix is too hot and PSR is far below the target.

Bit Monitor

The Bit Monitor is a valuable and versatile measuring instrument to monitor the digital activity not only of the audio file under investigation but also of the whole production chain by revealing problems in the digital signal.



[13] 24bit high resolution audio track with an effective
Bit Depth of 21bit for the left and 20bit for the right channel

1 Bit Ladder

The 64 bits of the audio signal are represented by a ladder of bit indicators with different intensities of petrol depending on the bit usage.

2 Bit Depth

This instrument applies a unique statistical measurement to determine the real bit depth of the audio track under investigation. Bits in a range from the 24th to the 17th bit that just contain pure noise are marked red.

The Depth check is especially useful to measure whether the analog to digital converter used for recording achieved the maximum possible bit depth. It is also an indicator for the quality of your analog signal flow.

Issues in the digital chain like plugins that process at 16 bits –truncating the signal– are easily detectable.

Resetting the measurement frequently improves the quality of the statistical data.

The example on the previous page shows a 24 bit high resolution audio track where the petrol numbers indicate the real bit depth of 21 bit (L) and 20 bit (R).

3 Bit Pattern

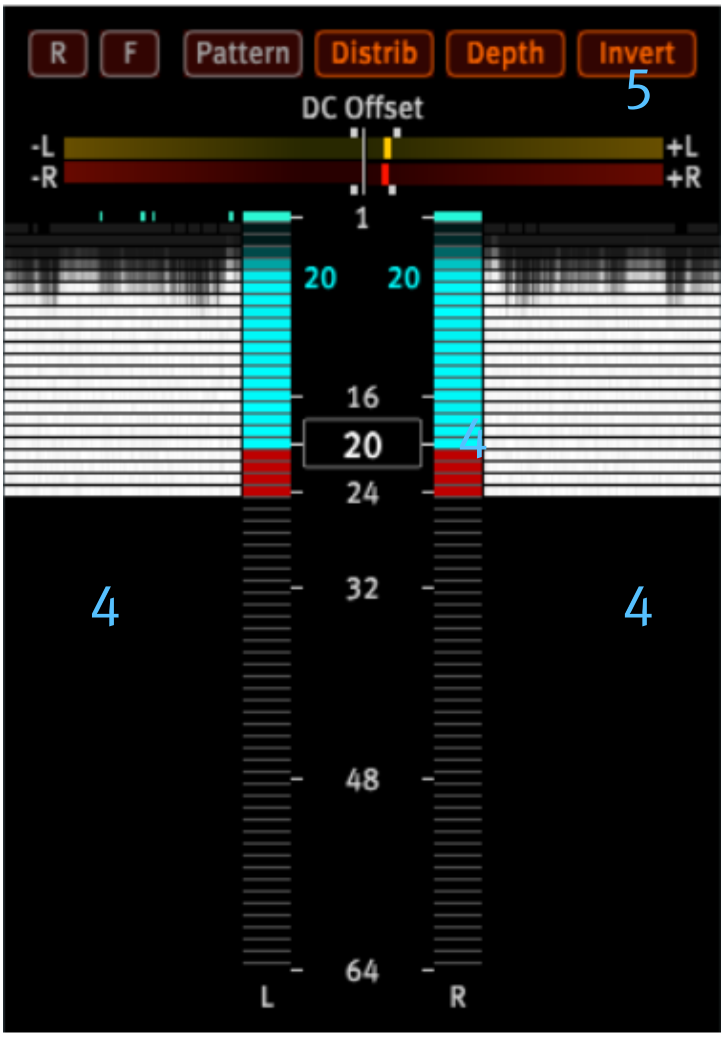
The Bit Pattern shows which bits are being used in a digital signal. Every 50ms an audio sample is decomposed in its bits that move outwards as a pattern to record a history of the bit usage. This is a more detailed view of bit usage compared to the brightness of the petrol bit ladder indicators.

A well recorded audio track should show a decreasing bit usage from the lowest to the highest bit. A regular bit pattern is an indication that something is probably wrong within the audio processing chain.

The top bits may not light up because the maximum level of the music is below a certain level (e.g. lower than -6dB). Up-scaled tracks (e.g. 16 to 24 Bit) can be identified by either the lower bits staying at zero or a bit pattern that is symmetrical around a certain bit number.

4 Bit Distribution

The Bit Distribution is an alternative display of the Bit Pattern. It shows the statistical commonness of bits by a gray scale where frequently used bits are indicated white. Unused bits stay black and rarely used bits are displayed in 50 shades of grey.



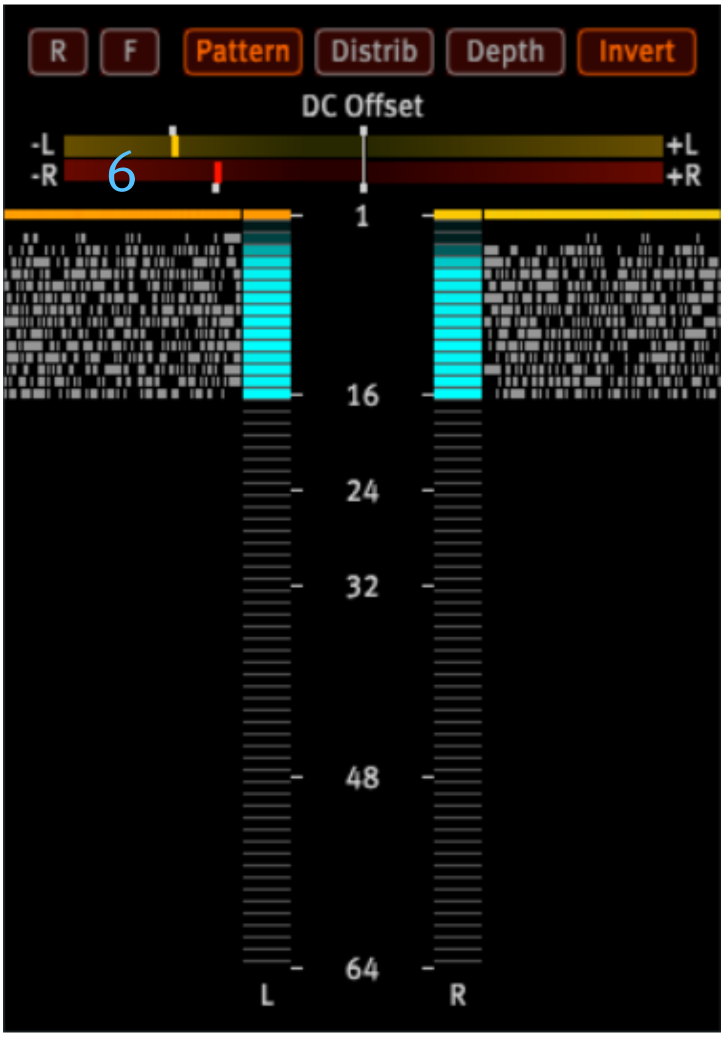
[14] Bit Distribution view of a 24bit high resolution audio track having a real Bit Depth of only 20bit.

5 Invert

The standard bit scale counts from the lowest (1) to the highest (64) bit. The scale can be inverted to make counting active bits easier (see Bit Depth).

Placing the mouse pointer above a bit ladder indicator reveals the number of the bit.

6 DC Offset



[15] Negative DC-Offset on the left and right channels

The top bit (sign) of the bit ladder indicates the strength of the DC offset separately for the left and right channels.

If the measured DC level is between -60dB and -40dB the top bit shifts its color from petrol over green and yellow to red. The history of the top bit is displayed in an outwards moving color pattern.

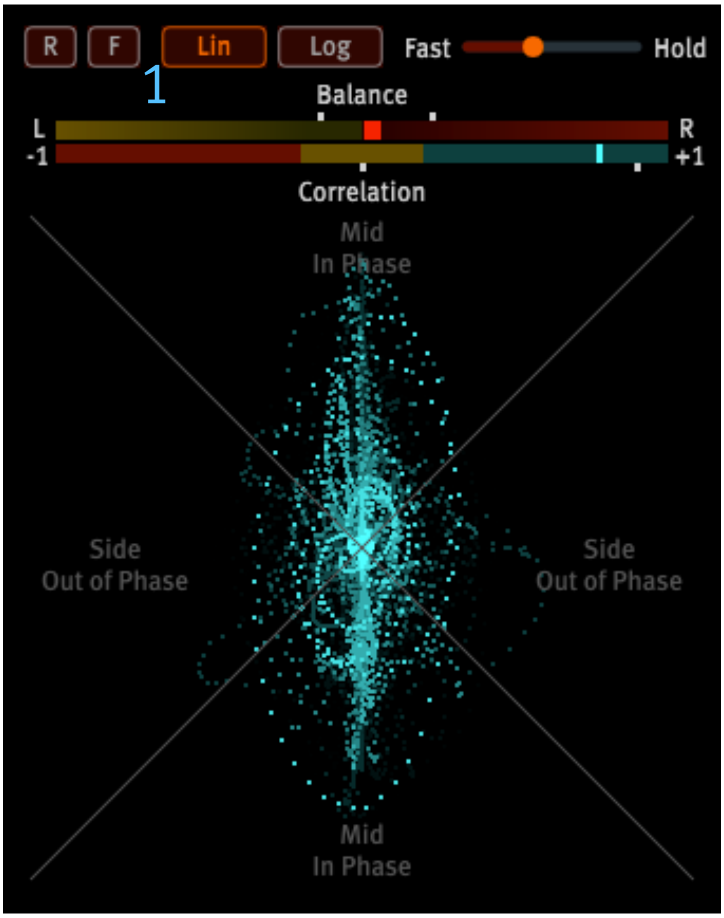
The additional DC meter distinguishes between a negative and positive DC offset, where the horizontal bars cover a range of up to -40dB of the full-scale signal.

The cause for DC is almost always created in the analog domain by faulty or bad gear.

DC should be avoided to mitigate issues like reduced headroom, problems with lossy audio formats or even power amplifiers going into protection mode.

Vector Scope

1 Linear Vector Scope



[16] Linear view of the audio vector (L & R)

The linear X-Y plot of the Vector Scope (Goniometer) reveals details about the relationship between the left and right channel, providing:

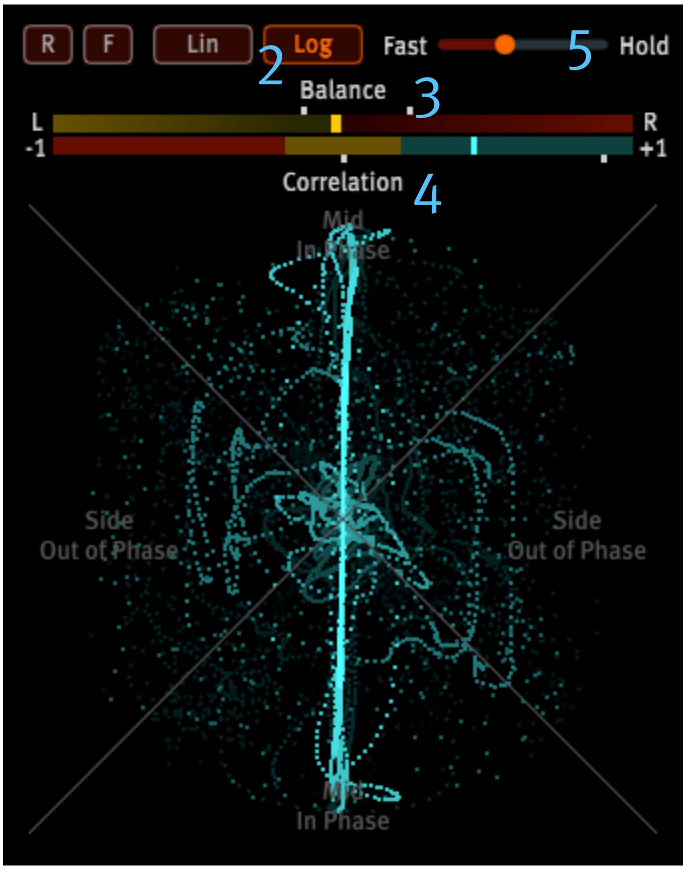
- a good idea about the audio levels for each channel
- the stereo balance and width
- its compatibility as a mono signal

With Lin engaged the relationship between the stereo channels is displayed linearly and has a maximum shape of a rhombus. The left channel drives the 45° deflection and the right channel drives the -45° deflection.

A monophonic signal, consisting of identical left and right signals, results in a vertical line, whereas any stereo component is visible as a deviation from this line.

A horizontal line indicates that the left and right channels are 180° out of phase.

2 Logarithmic Vector Scope



[17] Logarithmic view of the audio vector (L & R)

With Log engaged the relationship between the stereo channels is displayed logarithmically. This display is useful for smaller signal amplitudes and has a maximum shape of a circle.

3 Balance

The Balance meter shows the drift of the stereo track to either side. It helps to identify whether the stereo signal is well balanced. Small white brackets hold the maximum values.

The Pano/Phase instrument as part of the Spectrum Analyzer displays the Balance in far more detail. In fact, it is capable of showing the panorama for 4096 frequency bands.

4 Correlation

The correlation meter indicates the degree of similarity between the left and right channels.

A moving bar shows the momentary value, whereas small white brackets hold the maximum values.

The correlation meter is important to check for mono compatibility or to check if a stereo microphone position would result in phase cancellation.

Petrol values are always acceptable, the yellow area is still OK, but the red area indicates that phase cancellations take place. In a mono playback this would mean that certain frequencies are not audible with the same level as in the stereo playback.

The Pano/Phase instrument as part of the Spectrum Analyzer displays the phase in far more detail. It is capable of showing the phase for 4096 frequency bands.

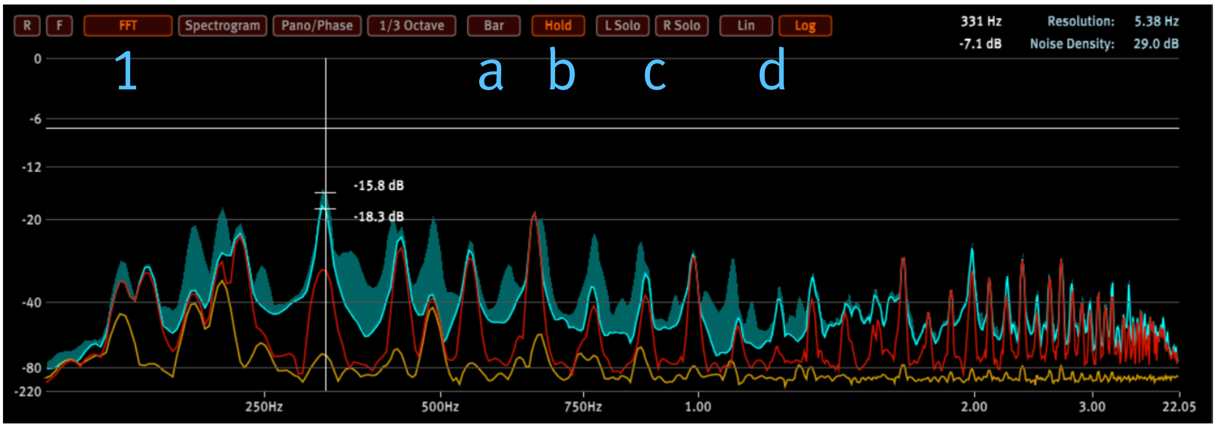
5 Luminescence Slider

The slider defines how long the Vector Scope signal stays on the screen. It is comparable to the luminescence of old cathode ray tubes.

If the slider is at its maximum the signal is held indefinitely.

Spectrum Analyzer

1 FFT – Fast Fourier Transformation



[18] FFT of L & R with maximum and falling envelopes to identify regions that e.g. need equalizing.

The FFT plots the audio spectrum displaying amplitude as a function of frequency.

The high speed FFT generates a new high resolution spectrum of 4096 frequency bands every 50ms (20Hz) covering a range from 0 Hz up to the Nyquist frequency (1/2x sample rate).

It achieves an exceptional frequency separation as well as amplitude precision by using a 7th order Blackman-Harris windowing function.

By placing the mouse cursor over the FFT window precise amplitude and frequency readouts are displayed.

The frequency resolution (e.g. 5.38Hz at 44.1kHz sample rate) and noise density (dB) are indicated by blue colored values.

Especially the noise density helps to determine the SNR (Signal to Noise Ratio) of the whole recording chain if measured in a frequency range that just contains noise.

On the Y-axis (vertical line) of the cross-hair two more values are being shown. The lower one is the falling true peak and the upper one is the true peak hold (envelope) of that specific frequency.

The display range of the amplitude scale (Y-axis) and logarithmic frequency scale (X-axis) can be stretched and compressed by mouse dragging.

Further functions available in FFT view are:

a) Bar

The FFT display can be switched between line and bar graph mode.

In bar graph mode the area beneath the envelope is filled with red bars for the right und yellow bars for the left channel.

b) Hold

When activating Hold a momentary peak hold line (light petrol) is introduced.
It displays the maximum values created by the left and right channel.

The petrol colored area above this line indicates the maximum peak hold values.

c) L Solo / R Solo

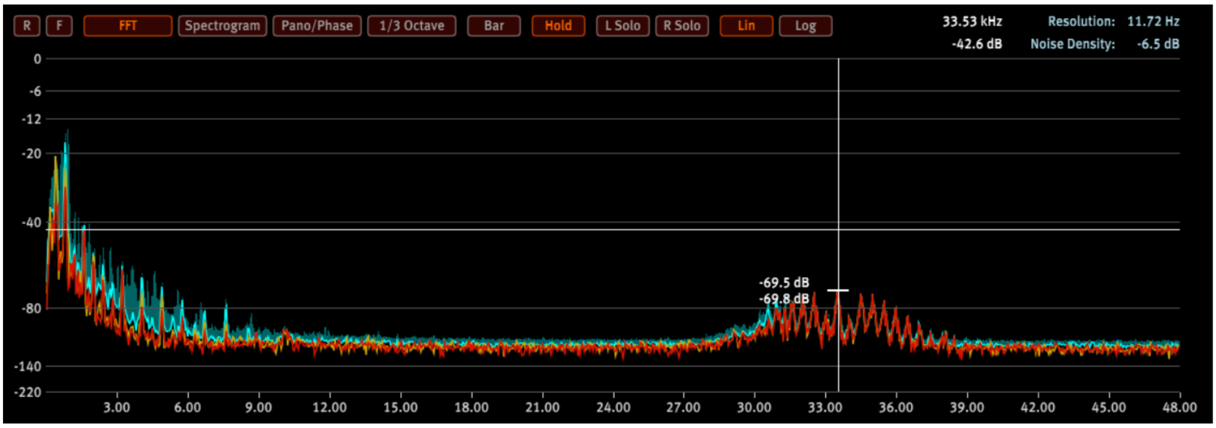
When activating L Solo or R Solo only the associated channel will be displayed.

d) Lin / Log

Switches the frequency scale (X-axis) to be linear or logarithmic.

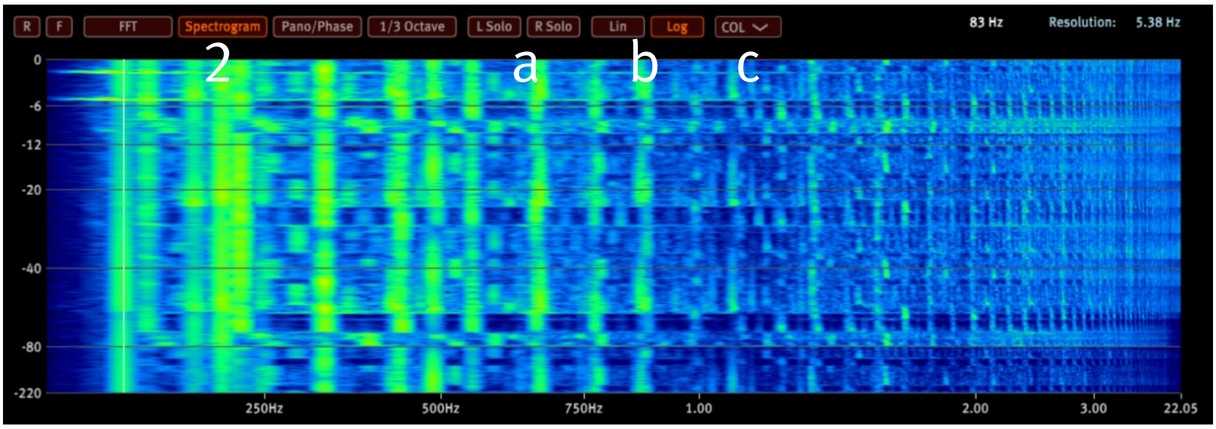
The linear frequency scale is fixed and is best being used to check high resolution content.

The logarithmic frequency scale should be used to check low frequency audio content. The X-axis can be stretched and compressed by mouse dragging.



[19]: Linear FFT of a 96kHz high resolution audio track revealing heavy distortions around 33.5kHz

2 Spectrogram



[10] Full color Spectrogram (waterfall plot)

The waterfall plot is built from high speed FFTs that are calculated every 50ms providing a resolution of 4096 frequency bands.

The Spectrogram achieves an exceptional frequency separation as well as amplitude precision by using a 7th order Blackman-Harris windowing function.

By placing the mouse cursor over the Spectrogram window, a precise frequency readout is given.

The FFT resolution (e.g. 5.38Hz at 44.1kHz sample rate) is indicated by a blue colored value.

Mouse dragging of the Y-axis stretches or compresses the dB scale which has a direct influence on the brightness of the waterfall plot.

The Spectrogram is an essential tool to detect issues like periodic distortions in the audio spectrum.

Further functions available in the Spectrogram view are:

a) L Solo / R Solo

When activating “L Solo” or “R Solo” only the associated channel will be displayed.

b) Lin / Log

The Lin and Log buttons switch the frequency scale (X-axis) to be linear or logarithmic.

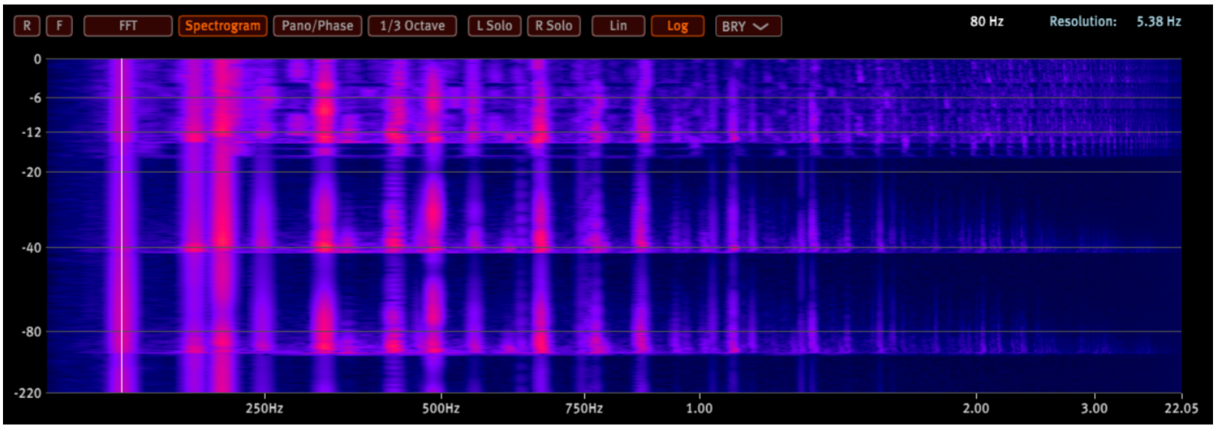
The linear frequency scale is fixed and is best being used to check high resolution content.

The logarithmic frequency scale should be used to check low frequency audio content. The X-axis can be stretched and compressed by mouse dragging.

c) Color scheme

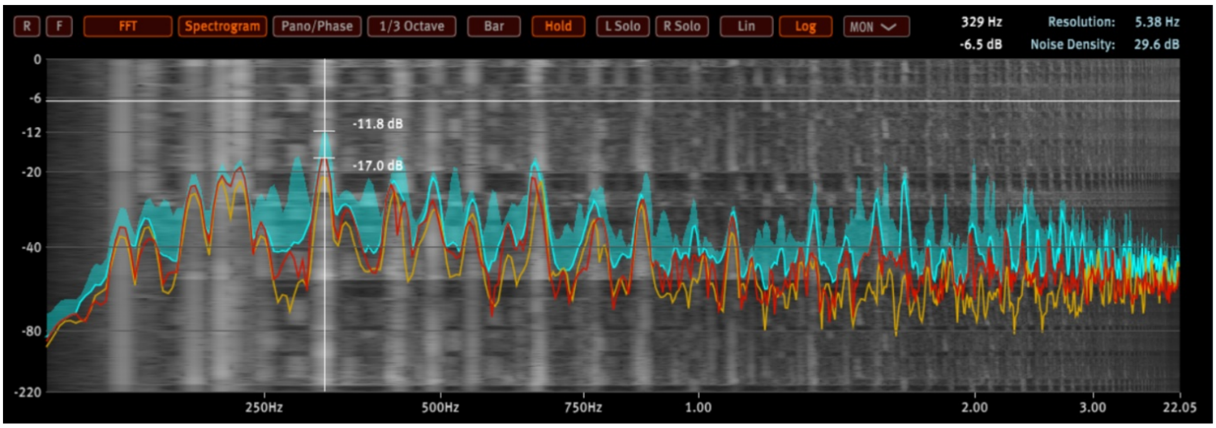
The drop-down menu offers a selection of color schemes:

BRY: Blue-Red-Yellow
COL: full color
MON: monochromatic gray scale



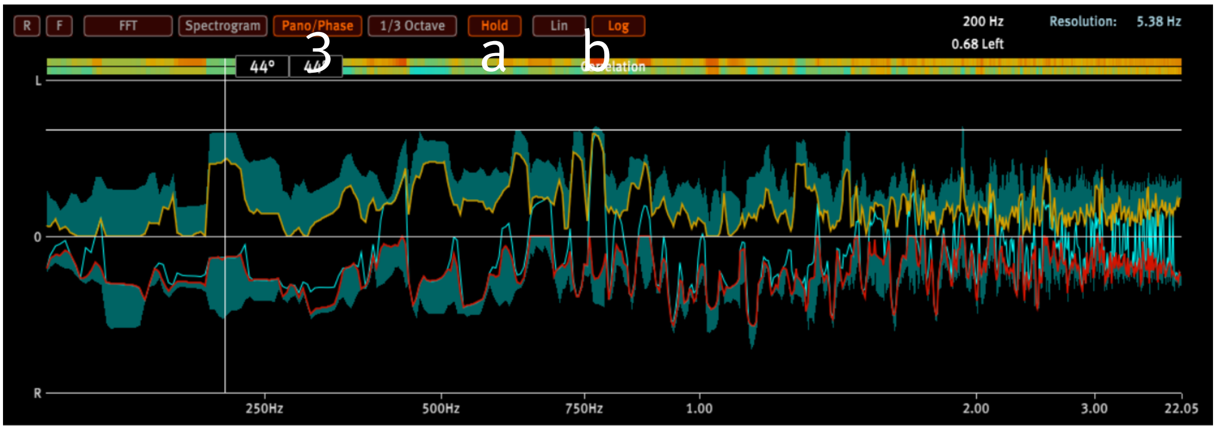
[21] Spectrogram with BRY color scale

The monochromatic scale is the best choice when layering the FFT above the Spectrogram.



[22] Monochromatic Spectrogram combined with FFT

3 Pano/Phase



[23] Hi-resolution analysis of Panorama and Phase

This instrument shows the precise panning to the left and right of 4096 frequency bands over a range of 0 Hz up to the Nyquist frequency (1/2 sample rate).

It complements the Balance display of the Vector Scope by showing the L/R panning over 4096 frequency bands.

Furthermore, the two Correlation stripes complement the Correlation meter of the Vector Scope by indicating the phase relationship of 4096 individual frequencies by applying a similar color coding as used by the Correlation meter.

The lower Correlation stripe shows the momentary correlation wheres the upper correlation stripe holds the negative correlation values over the time of measurement.

Stable red marked frequency ranges below 4kHz should get a deeper investigation because they are an indication that the left and right channel loose energy in those frequency bands by cancelling each other out. These cancellations are quite critical, especially in the area of bass frequencies.

Further functions available in the Pano/Phase view are:

a) Hold

When activating Hold two momentary peak hold lines (red and yellow) are introduced. They indicate the falling peak values of the left and right channel.

The petrol colored areas above these lines hold the maximum peak values.

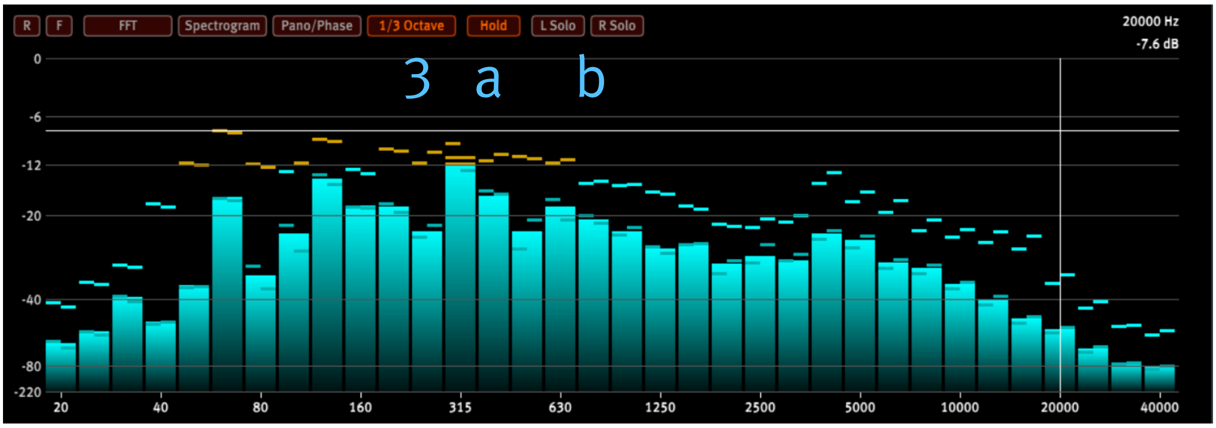
b) Lin / Log

Switches the frequency scale (X-axis) to be linear or logarithmic.

The linear frequency scale is fixed and is best being used to check high resolution content.

The logarithmic frequency scale should be used to check low frequency audio content and can be stretched and compressed by mouse dragging.

4 1/3 Octave Analyzer



[24] 1/3 Octave analysis of a 96kHz high resolution audio track

The analyzer shows the frequency spectrum in bars separated by 1/3 octave over a range of 20 Hz up to the Nyquist frequency (1/2 x sample rate).

The bars represent the mono amplitude while the falling peaks and peak hold values are represented as lines, separately for the left and right channel.

Mouse dragging of the Y-axis stretches or compresses the dB scale.

Further functions available in 1/3 Octave view are:

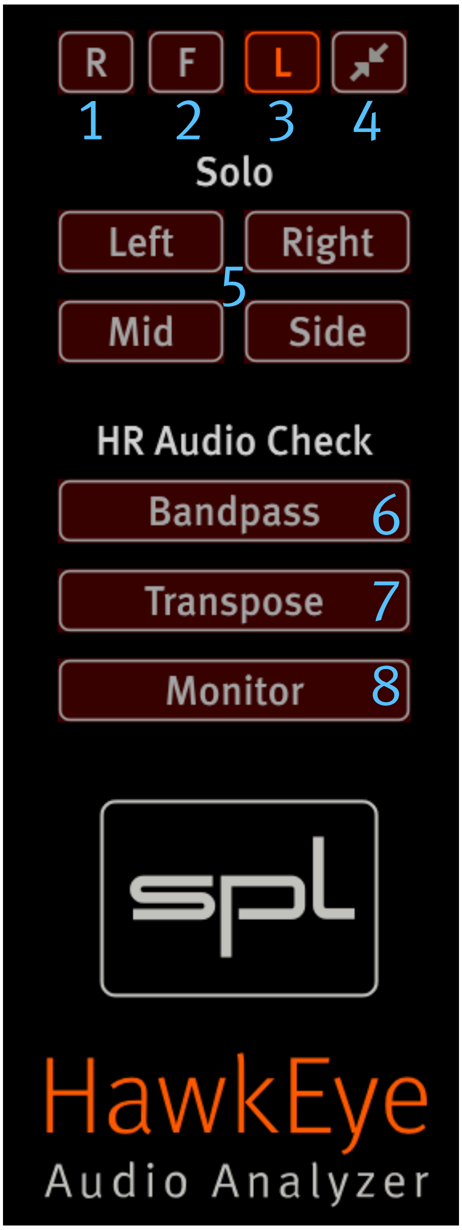
a) Hold

Activating Hold introduces peak hold lines for the left and right channel.

b) L Solo / R Solo

When activating L Solo or R Solo only the associated channel will be displayed.

Control Panel



[25] HawkEye Control Panel

1 R – Reset

Clears the calculations of all instruments and restarts all measurements.

2 F – Freeze

Freezes all displays while measurements continue in the background.

3 L – Labels

Activate Labels to add detailed labelling to certain instruments, scales and value boxes. It also enables the tooltips which pop up when the mouse cursor hovers over a button or value box.

4 Minimize/Maximize

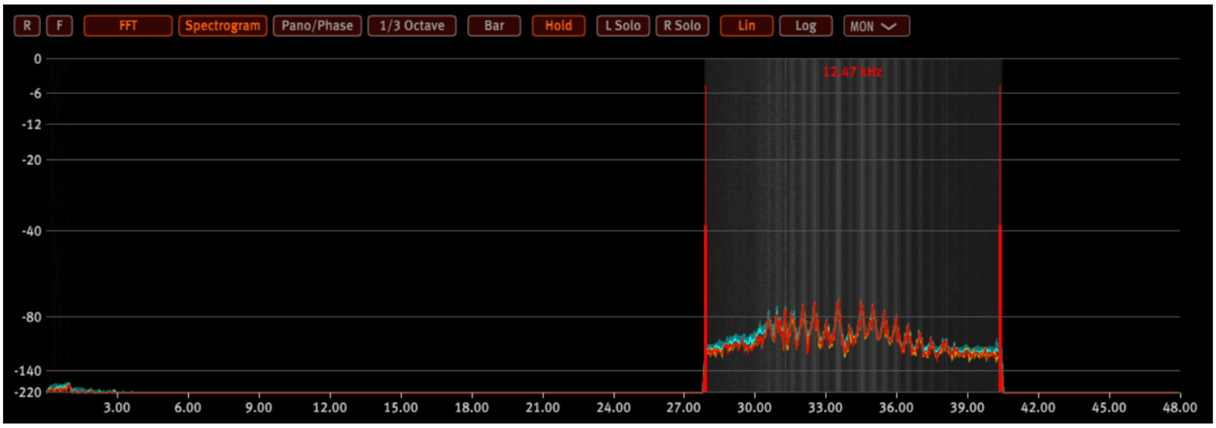
This function minimizes the main view to the Control Panel and another click expands it back to the full view.

Minimize the view to safe screen space and undock only those measuring instruments really needed by double clicking on them.

5 Left / Right / Mid / Side

These are solo buttons. All analyzing tools just work on the solo'd channel.

High Resolution Audio Check



[26] Bandpass isolated heavy distortion of a 96kHz high resolution audio track.
With Transpose those distortions are made audible.

6 Bandpass

Checking the frequency range beyond 20kHz for noise vs. audio is important when verifying real high resolution recordings. Therefore, HawkEye offers the Bandpass function to select certain frequency ranges for analysis.

Two red vertical lines in the FFT, Spectrogram and Pano/Phase view define the upper and lower cut-off frequencies, which can be changed individually or by moving the whole bandpass.

Furthermore, it is possible to set a new bandpass by just dragging the mouse over the target frequency range. It is recommended to use the linear frequency scale.

7 Transpose

Checking high resolution audio content for noise vs audio is hardly possible if you are not a bat. Therefore, HawkEye offers a frequency shift function that lets you listen to audio content beyond the human auditory capabilities by shifting the selected frequency band into an audible frequency range.

[27] Bandpass isolated distortion shifted into the audible frequency range

8 Monitor

You need to activate Monitor to listen to the transposed frequencies or the solo'd signals (Left, Right, Mid, Side).

If Monitor is not engaged the named functions are only visible. When Monitor is engaged audio will be routed through HawkEye and is processed.

ATTENTION: Lower the volume before switching off Bandpass!

You may have listened to the selected high frequency content at a much higher volume. Going back to full range listening by switching off Bandpass without reducing the volume may lead to an extremely loud playback that may damage your speakers or even your hearing. Be careful.

CPU-Load Considerations

HawkEye analyzes a lot of audio features in parallel and normally works on modern computers without any performance issues. Nevertheless, to support systems with lower CPU performance we like to suggest methods to reduce the system load:

- HawkEye renders only graphical content that is visible. If only the Level Meter is required and active, undock it and minimize the main window (see Control Panel 4). By doing so the CPU load goes down because of a heavily reduced rendering effort.

- Freezing single instruments reduces the overall CPU load.

- Choose a sensible buffer size between 10ms and below 50ms.

All settings can be stored as pre-sets to be re-called at any time.

Stand-alone Version (OS X and Windows)

[28] HawkEye program icon

HawkEye is additionally distributed as a stand-alone version for OS X and Windows.

The following use cases are covered:

1. Analyzing the audio input to use HawkEye as metering solution outside the DAW.
2. Re-Route the audio output of a streaming client or media players to check its quality.

Especially for the second case we need to use a virtual sound card to route the audio between different applications.

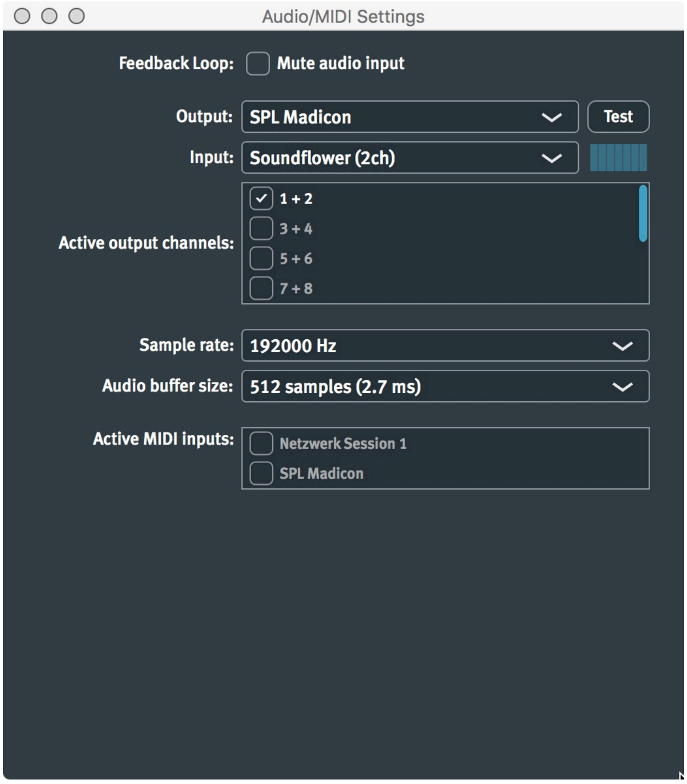
As of our knowledge there are currently the following free solutions available:

OS X: Soundflower

It is easy to use and presents itself as an audio device in the Audio/Midi Setup, allowing any audio application to send and receive audio.

Windows: VB-Cable Virtual Audio Device

Audio/MIDI Settings



[29] Audio/MIDI Settings

To configure the audio settings of the standalone version, click on Options in the upper left corner:

- Select for example Soundflower as Input and your output device as Output.
- Select the output channel pair to where the input channels shall be routed in the Active output channels selection field.
- Clicking on Test sends a 400Hz one second test tone to check the connection.
- There is the risk of a feedback loop in case of an identical Input and Output device. Click the “Mute audio input” check box to avoid feedback loops.
- The Sample rate selection determines the frequency range of the Spectrum Analyzer (FFT, Spectogram, Pano/Phase and 1/3 Octave Analyzer).
- The Audio buffer size is best set to values between 10ms and below 50ms to have an immediate fluent visual reaction to the actual audio playback. Lower settings may lead to hiccups. Deactivating the “Monitor” function or keeping the buffer sizes above 10ms mitigates those issues. The performance of your computer system determines how fast you can go.

HawkEye Audio Analyzer

Version 1.0

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Programming by Stephan Hotto