MusicScope Manual

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Compatibility

The MusicScope is Windows (7 – 8.1 / 32 & 64 Bit) as well as Mac OS X (10.7.3 – 10.10.x) compatible.

Operations

The MusicScope has been designed with focus on operational simplicity.

A music track can be loaded via:

1. Load (Folder Icon) – Opens a file dialog to load one or several tracks
2. Drag & Drop – Load one or multiple files by dragging them into the GUI
3. Playlist – Drag & Drop several audio files into the Playlist to play or analyze them without further intervention.

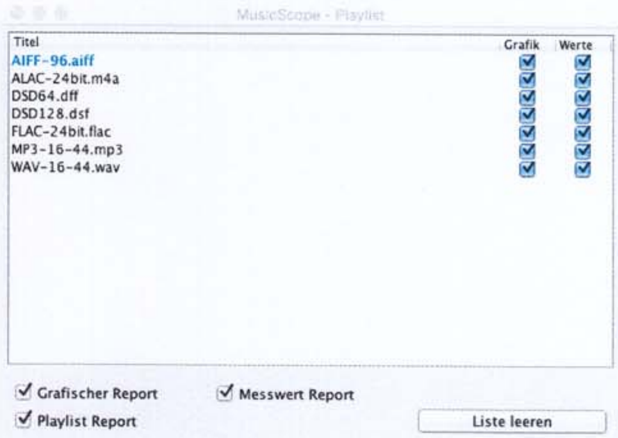
The Playlist supports the playback and analyzing of several audio tracks and opens automatically as soon as multiple files are loaded. The list can be opened at anytime by clicking on the track name. By "Drag & Drop" tracks can be added or moved up and down in the list.

A context menu allows the deletion of single items, whereas "Clear" resets the whole Playlist. It is possible to select different report types for each track or even an overall report for the whole list.

Start the analyzing process:

Play starts the playback, whereas Analyze (Microscope Icon) starts the fast analyze mode. It is possible to switch anytime between Play and Analyze to go quickly through the track or to start the audio monitoring of dedicated parts.

A Pause-Function allows a hold of the analysis at any time to have a look at the current measurement results.



Realtime Analysis

The Audio Input Mode allows the selection of an Audio Input Device for Realtime Analysis:

a.) Open the Settings Menu (gear-wheel icon)

b.) Select the Audio Input Tab

c.) Select the Audio Input Device as well as the sampling rate and bit depth

d.) Click on Select

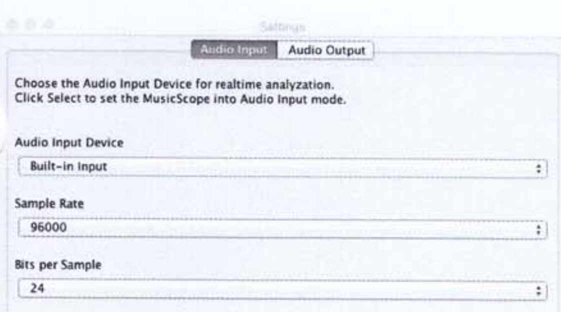
e.) Click on „Play" starts the Audio Input

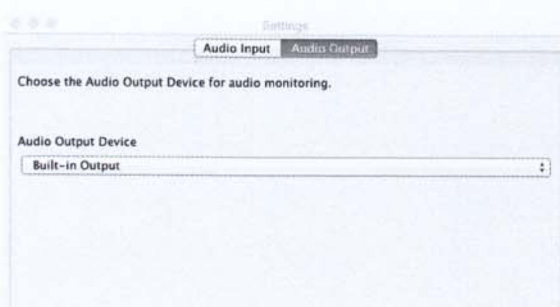
Applications for the Realtime Audio Analysis:

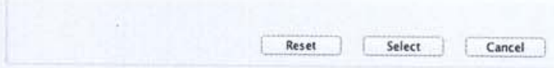
– Vinyl-Record analyzing via DA-Converter or phono amplifier connected to the line-in of the audio device.

– Analyzing streaming services by routing their output via virtual audio devices into the MusicScope.

The Audio Output dialog provides the settings for the output device to be used for audio monitoring. To prevent audio feedback under dedicated configurations it is possible to set the output to „No Audio Output".







Format

The MusicScope handles several lossless and lossy audio formats:

+ WAV – Waveform Audio File Format (WAVE)

+ BWF – Broadcast Wave Format

+ AIFF – Audio Interchange File Format

+ FLAC – Free Lossless Audio Codec

+ ALAC – Apple Lossless Audio Codec

+ DSD (DFF & DSF) – Direct Stream Digital

+ DSDIFF (by Philips) / DFF = Direct Stream Digital Interchange File Format

+ DSF (by Sony) = DSD Stream File

+ MP3 – MPEG-2 Audio Layer III

+ Supported sample rates (kHz) are: 44.1, 48, 88.2, 96, 176.4, 192, 352,8, 384 as well as DSD64, DSD128 & DSD256

+ Supported Bit depth: 1, 16, 24 and 32 Bit

The audio monitoring (playback) of the MusicScope supports material of up to 24 Bit / 384 kHz on Mac OS X systems. Direct Stream Digital (DSD) can be reproduced without using a special DSD capable Digital to Analog Converter (DAC).

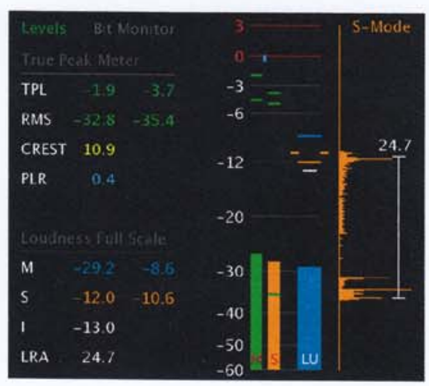
Left/Right & Mid/Side Analysis

Beside of analyzing the Left & Right audio channels it is possible to switch into a Mid/Side mode. The Mid-Signal represents the mono and the Side-Signal the stereo part of the audio signal.

Analyzing in Mid/Side mode provides detailed information about the stereo image and therefore the quality of the stereo reproduction.

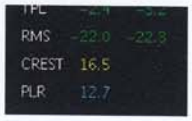
A single mouse click on the graphical levels display switches between Left/Right and Mid/Side mode. In M/S mode the labeling and coloring of the peak bars changes.

In Mid/Side-Mode the history circle shows the Mid & Side peaks instead of the L+R peak and Loudness in Left/Right mode.



Levels

True Peak Meter:



TPL:

The True Peak Level represents the signal amplitude after the digital to analog conversion to detect Inter Sample Peaks. The simulation of the analog domain is possible because the meter is doing an up-sampling of the measured input signal to interpolate the waveform. A green value and green bars (graphical representation) below 0 dBFS (Decibel Full Scale) indicate a good leveling whereas red marked values clearly show Inter Sample Peaks, which could cause distortions.

RMS:

The value displays the 400ms averaged RMS (Root Mean Square). Light green bars within the filled peak bars represent the RMS in the graphical display. To calculate the RMS the mathematical definition has been used where a 0 dBFS sinus shows an RMS of -3 dBFS and therefore a CREST of 3 dB.

CREST:

The crest factor is the relation between the Peak und RMS values in dB. A pure sinus signal has a crest of 3 dB whereas a square wave shows 0 dB. Heavy compressed and limited music can reach values below 4 dB. A good native studio master should have a maximum crest larger than 10 dB.

PLR (Experimental):

The PLR is the relation Peak to Loudness in dB. and an indication for the momentary dynamic.

Loudness Full Scale:

The EBU (European Broadcast Unit) and ITU (International Telecommunication Unit) defined the standards EBU R128 and ITU-R BS-1770 that are fully implemented by the MusicScope. The algorithm to calculate the loudness considers the physiology of human hearing and therefore delivers a much better indication of the perceived loudness than the RMS.

The loudness values differ in their integration (averaging) times:

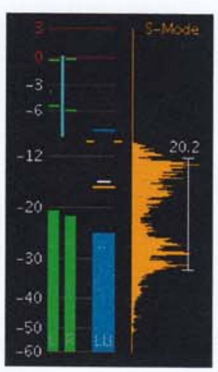
Momentary Loudness (M) = 400 ms

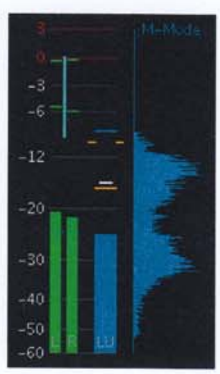
Short Term Loudness (S) = 3 s

Integrated Loudness (I) = The averaged loudness over the whole track

Loudness Range (LRA) = This is a very good representation of the music track dynamics

All Loudness values are available as numerical values and as a graphical display. The M- and S-Loudness maximum values are displayed as well.





Loudness Histogram:

The vertical Loudness Histogram indicates, similar to a photo histogram, the distribution of Momentary Loudness (M-Mode) or Short-Term-Loudness (S-Mode) and is therefore a further measurement, beside of the LRA, to assess the track dynamic. A click on S-Mode or M-Mode switches between the different modes. The S-Mode provides a graphical display of the LRA to make the value more tangible.

Bit Monitor / DC-Indicator

The Bit Monitor displays the bit usage. Unused bits are marked blue, whereas different shades of gray indicate the bit utilization. A rolling Bit History makes it easy to identify stuck bits or regular bit pattern, indicating up-sampling or faulty recording software or hardware.

A Direct Current Indicator helps to identify any DC offset.

A mouse click resets the measurements.

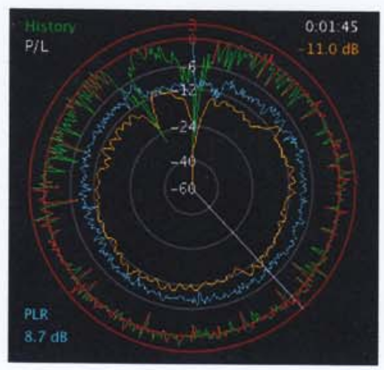
History

The circular diagram shows the evolution of the peak values (green), Short-Term-Loudness (orange) as well as the Peak over Loudness Ratio (blue) for the whole music track.

Red peaks are caused by inter-sample peaks, exceeding 0 dB. Frequent inter-sample peaks are a source for audible distortions.

It is possible to use the mouse pointer to measure the S-Loudness and PLR for a specific track time.

In Mid/Side mode the Histogram shows the peaks for the mid (mono) and side (stereo) signal.



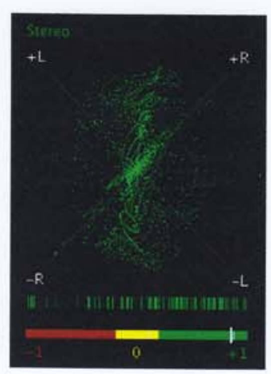
Stereo Meter

The Stereo-Meter is divided into three parts and provides the means to get a better understanding of the stereo image.

1.) Vector Scope – Distribution of the Stereo-Field.

2.) Balance Indicator – An indication of the current signal position in the stereo plane and its width.

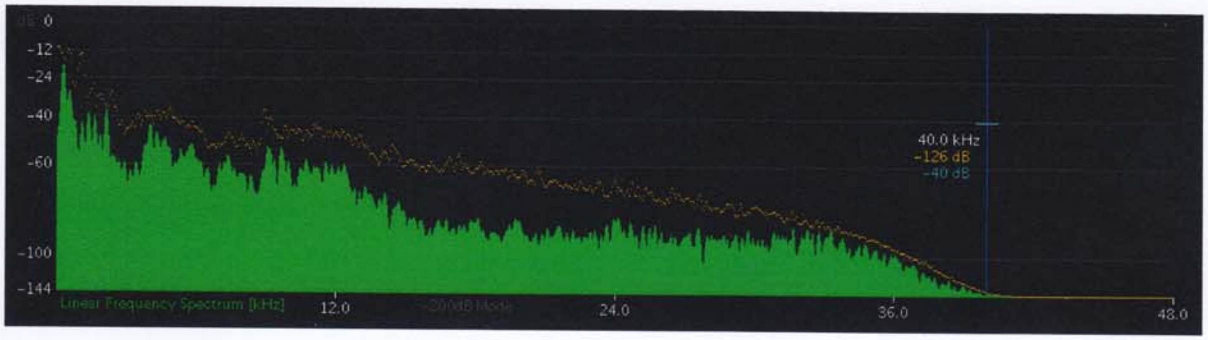
3.) Correlation Meter – Green values show mono compatibility and indicate a good localization within the stereo image, because the de-correlated signal part is quite low.



Linear and Logarithmic Frequency Spectrum

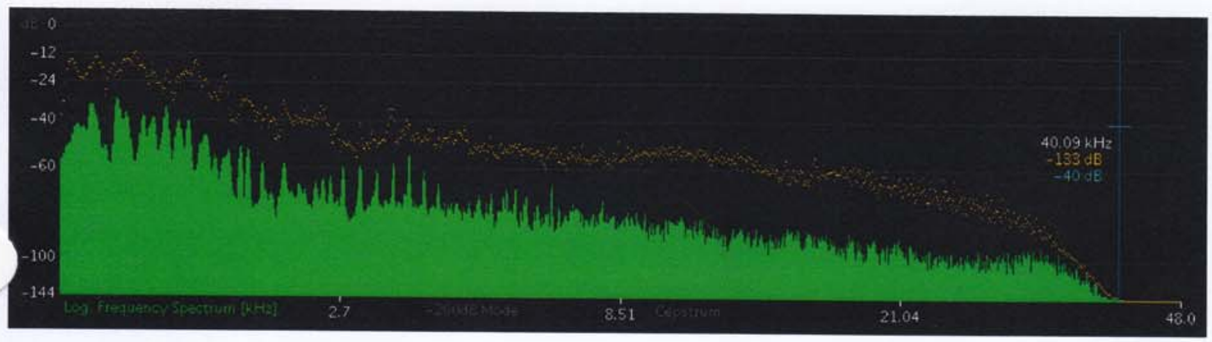
The frequency content of the music can be displayed within a linear or logarithmic spectrum.

A linear representation is especially interesting for the measurement of high-resolution audio files.



The amplitude scale in decibel (dB) can be changed by vertical mouse dragging.

The logarithmic Spectrum provides a higher resolution for the lower frequencies.



In case of the logarithmic spectrum the frequency scale can be adapted via horizontal mouse dragging on the scale.

To extend the amplitude resolution a „-200dB Mode" is available.

Cepstrum

The Cepstrum provides in-depth information about the periodicity of the spectrum and therefore about harmonics within the original signal.

To activate the Cepstrum-Display it is necessary to switch the Spectrum into the logarithmic frequency scale. A Cepstrum switch enables the activation of the function.

The values of the Cepstrum do not correspond to the scales of the Spectrum. The mouse pointer is used to fetch the Cepstrum's Quefrency in ms (milliseconds) and kHz.

Areas of Application:

– Identifying harmonics in music.

– Finding very small amounts of harmonics in sinus signals used for analysis (e.g. in-depth Sample Rate Converter Analysis).

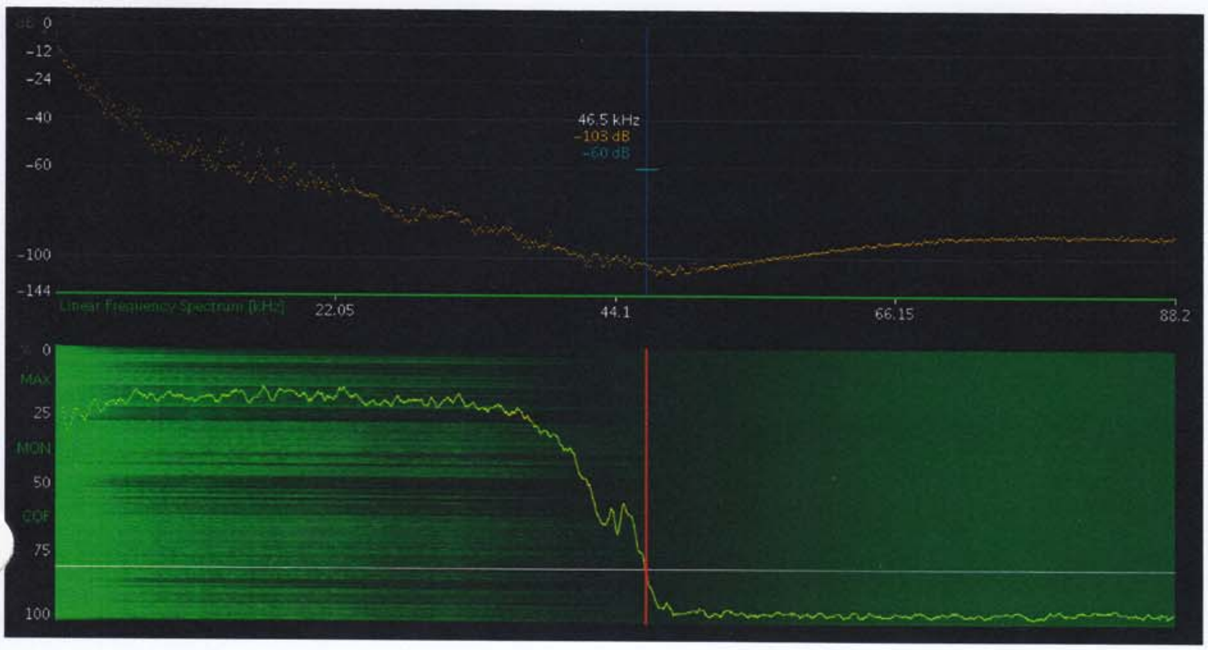
– Reveal mechanical errors within gear boxes, engines, etc. by an acoustic analysis.

The large red peaks of the Cepstrum screenshot are 1.13ms (around 880Hz) apart, indicating that there are harmonics of 880Hz in the spectrum. The mouse pointer displays the 880Hz fundamental frequency within the spectrum.

Spectrogram

The Spectrogram is a representation of the spectrum over time and enables the detection of periodical interfering signals as well as the determination of the highest frequencies containing music to analyze High Resolution Audio tracks.

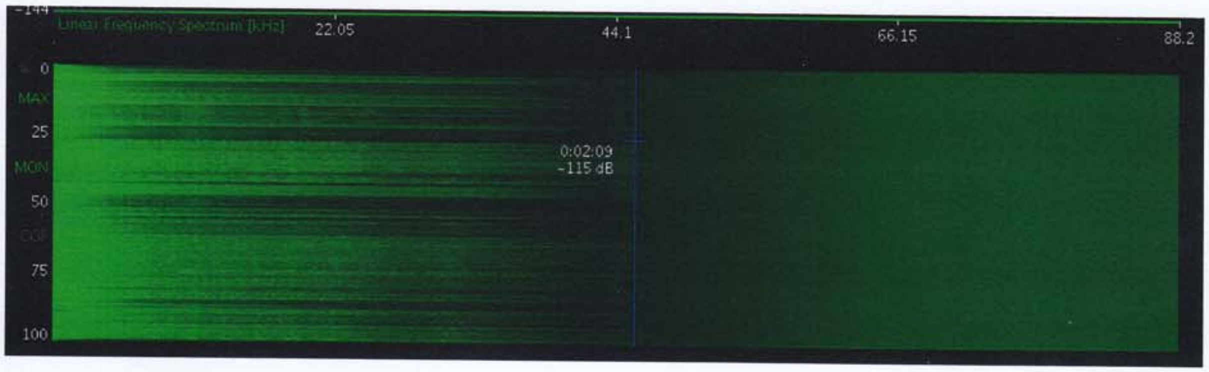
The amplitude spectrum is represented by the intensity of monochrome green or as color range. Hovering the mouse pointer over the spectrogram displays the absolute values in decibel as well as the track time below the mouse pointer. A click on "MAX / AVG / MIN" toggles the spectrogram between maximum, averaging and minimum mode.

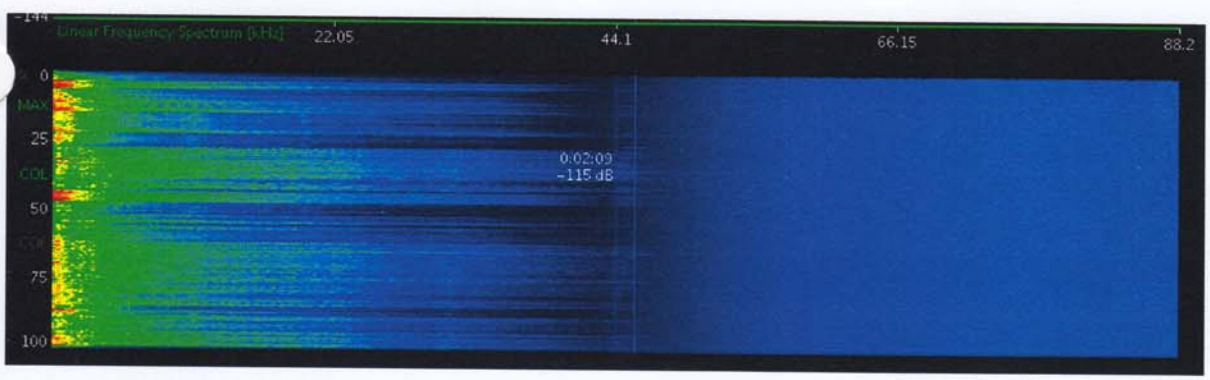


An automatic cut-off frequency detection algorithm makes it possible to estimate the frequency range that contains music. The detection becomes active if a high-resolution audio file with at least 88.2 kHz sample rate has been loaded and the COF switch (left side of the Spectrogram) is activated.

If the algorithm is not able to calculate the cut-off frequency then the COF-Switch changes to red to indicate an error. This is always the case for sample rates below 88.2 kHz and audio files with complex spectrograms, where a manual analysis is recommended.

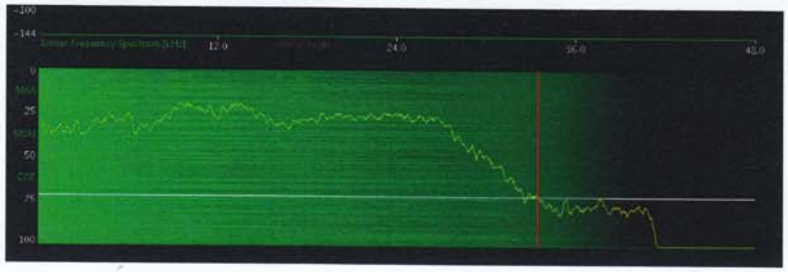
The automatic detection can be wrong. The best way is to verify whether the yellow curve falls clearly and consistently below the white threshold, indicating that there isn't any music signal within the associated spectral range.





Reporting

At the end of the music track analysis the measurement results stay in the display for further investigation. To save those data a reporting function exports them as picture (.png) and/or text file (.txt). A click on the "Report" button, which replaces the "Track Time" display at the end of a full analysis, opens the report save dialog.



```
DSD128.dsf_report.txt
Report generated by the MusicScope - www.xivero.com

File:                     DSD128.dsf

Audio format:               DSD
Bit depth:                    1 bit
Sample rate:             2822.4 kHz

TPL Left:                  -2.5 dB
TPL Right:                 -1.3 dB
RMS Left:                 -18.8 dB
RMS Right:                -17.7 dB
Crest Min.:                 3.0 dB
Crest Max.:                10.6 dB

Max. M-Loudness:           -9.9 dB
Max. S-Loudness:          -11.5 dB
Integrated Loudness:      -15.5 dB
Loudness Range:             6.7 dB

Spectrum:
[kHz]                       [dB]
1                           -15.2
2                           -21.7
3                           -29.2
4                           -34.1
5                           -31.9
```

The Playlist provides similar report settings. If the tick box for the "Overall Report" is selected then a file save dialog opens at the end of the Playlist-Analysis to choose the destination for the report.

test.txt

Report generated by the MusicScope - www.xivero.com

Track	Audio format	Bit depth	Sample rate	TPL Left	TPL Right	RMS Left	RMS Right	Crest Min.	Crest Max.	Loudness Range	Integrated Loudness
AIFF-96.aiff	PCM	16	44100 Hz	-1.9 dB	-1.9 dB	-4.9 dB	-4.9 dB	3.0 dB	3.0 dB	0.0 dB	-2.0 dB
ALAC-24bit.m4a	PCM	24	88200 Hz	-10.5 dB	-9.9 dB	-26.6 dB	-26.6 dB	6.0 dB	10.1 dB	4.5 dB	-23.2 dB
DSD128.dsf	DSD	1	2822400 Hz	-2.5 dB	-1.3 dB	-18.8 dB	-17.7 dB	3.0 dB	10.6 dB	6.7 dB	-15.5 dB
FLAC-24bit.flac	PCM	24	88200 Hz	-10.5 dB	-9.9 dB	-26.6 dB	-26.6 dB	6.0 dB	10.1 dB	4.5 dB	-23.2 dB
MP3-16-44.mp3	PCM	16	44100 Hz	-10.5 dB	-9.9 dB	-26.6 dB	-26.6 dB	6.1 dB	10.7 dB	4.5 dB	-23.2 dB
WAV-16-44.wav	PCM	16	44100 Hz	0.2 dB	0.1 dB	-16.4 dB	-17.0 dB	0.3 dB	11.7 dB	3.2 dB	-14.4 dB

MusicScope Use Cases

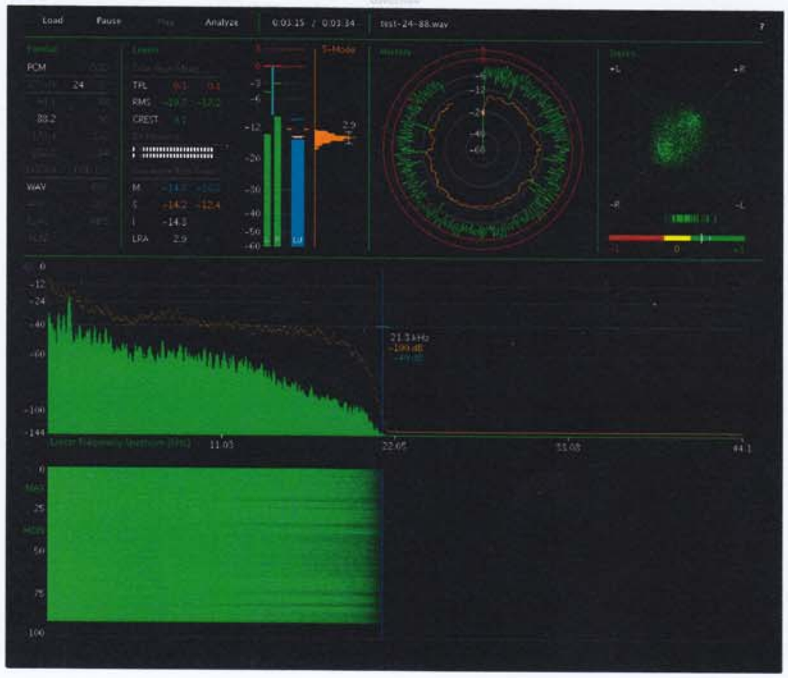
- Faked High Resolution Audio

We want to analyze a high resolution audio track where we have the suspicion that it is just an up-sampled version of a CD source.

The Format Display tells us that the audio file contains data in the format of 24 Bit / 88.2 kHz. This would mean that the highest frequencies could be 44.1 kHz (½ x sampling frequency / see Nyquist).

If we check the spectrum and spectrogram then we see that the music just contains frequencies up to around 21 kHz. The steep frequency decline around 21 kHz and spectrogram values mostly above -100 dB (mouse pointer measurement) indicate that this was originally a 16 Bit (max. -96dB) recording.

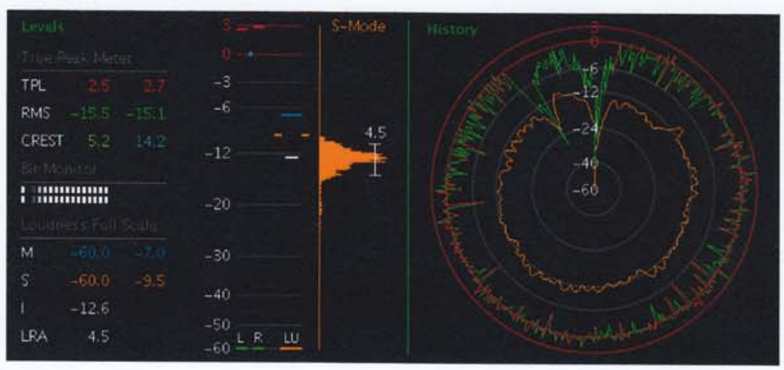
We simply proved that this was a CD recording with 16 Bit / 44.1 kHz, up-sampled with dedicated software, pretending being a high resolution audio track.



Inter Sample Peaks

The recording shows several Inter Sample Peaks indicated by TPL values of about +3 dBFS and a History Display with red marked overs of up to around +3 dB.

Theoretically it isn't possible in a digital system to have higher levels than 0 dBFS, but if the signal is converted back into the analog domain then those Inter Sample Peaks cause distortions because the headroom of the converter is not large enough to reflect the resulting amplitudes.

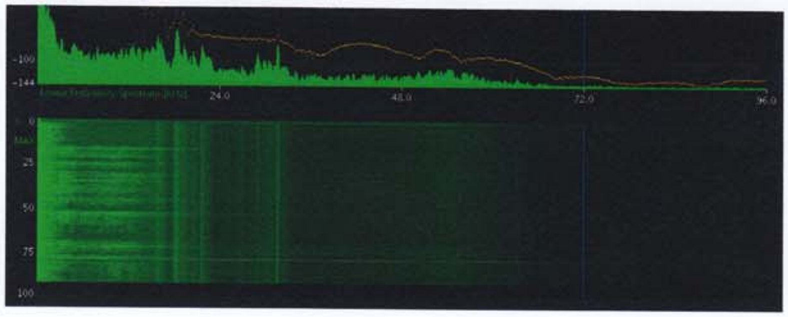


Periodical Interfering Signals

Periodical distortions are easily detectable within the spectrum and spectrogram.

The image below shows several vertical lines in the spectrogram and peaks within

lines in the spectrogram and peaks within the spectrum indicating interfering signals. The analyzed record is a digitalization (24 Bit / 192 kHz) of a studio master tape.



Stereo Image Quality

The MusicScope provides several means to asses the quality of the stereo image.

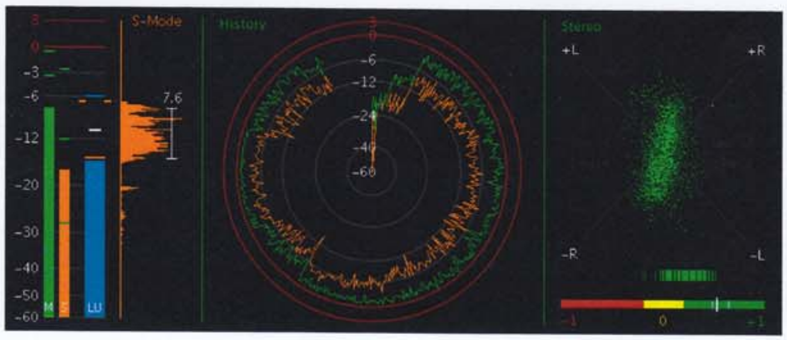
1. Stereo-Meter
   a. Vector Scope
   b. Balance Indicator
   c. Correlation Meter

2. Mid / Side Mode

This is a special analyzing mode activated by a single mouse click on the Levels-Meter. It allows the separate measurement of the mid (mono) and side (stereo) information. Within Mid/Side-Mode the History displays the mid (green) and side (orange) signal. A good Stereo Image allows an easy localization of instruments and voices.

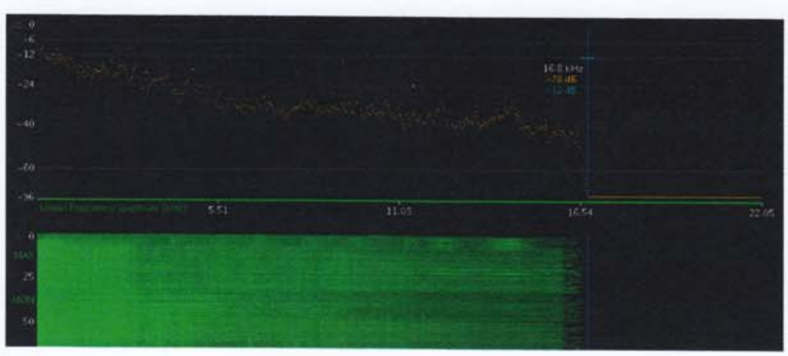
If the following conditions are met then these are a good indication for an acceptable stereo reproduction:

- The Vector Scope shows a graphic which has a more vertical than horizontal distribution.
- The balance indicator is most of the time around the mid of the display and not too wide.
- The Correlation Meter stays in the green area
- In Mid/Side Mode the History indicates that the mid signal is always larger than the side signal.

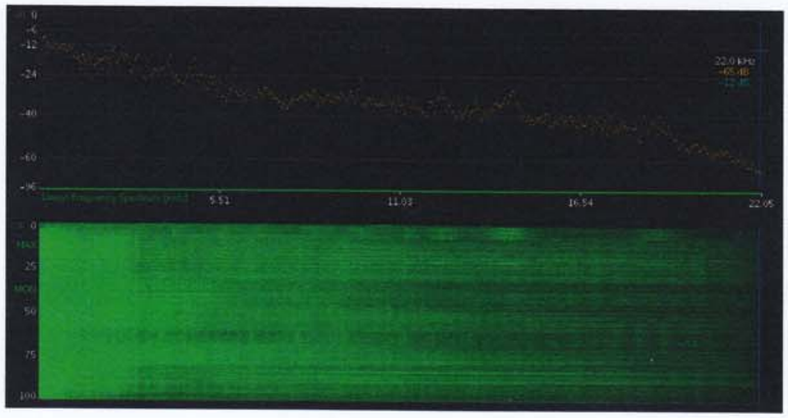


MP3 versus uncompressed audio

This is a comparison between an uncompressed and a MP3 (128 kBit/s) version of the same audio track. The first Spectrogram shows that the MP3 version is heavily frequency limited and that parts of the music have been removed on basis of a psychoacoustic model.



Original WAV version:

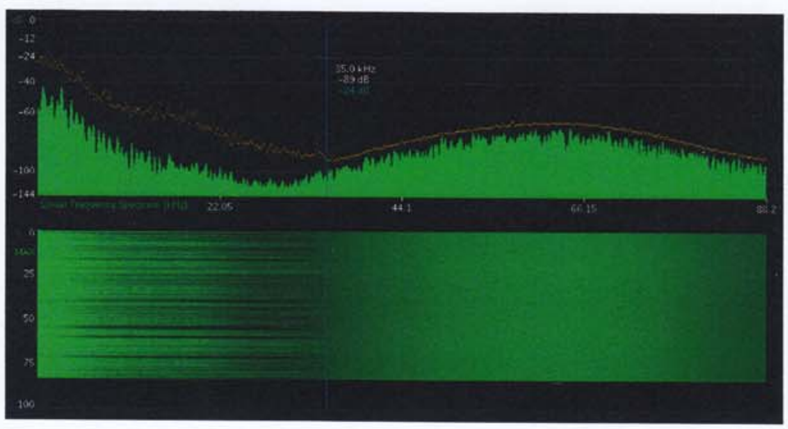


- Direct Stream Digital (DSD) Analysis

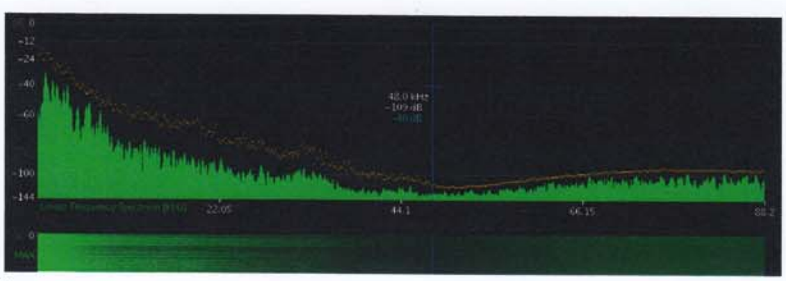
The following two examples represent a DSD64 (1 Bit / 2.8224 MHz) and a DSD128 (1 Bit / 5.6448 MHz) audio track.

The MusicScope is able to analyze and to playback DSD tracks without the need for a dedicated DSD Digital to Analog Converter (DAC).

The analysis of the DSD64 track indicates that there is still music content around 35 kHz which goes over into quantization noise given by the 1 Bit digitization method.



The DSD128 track moves the quantization noise to higher frequencies. The spectrogram confirms frequency parts of the music up to 48 kHz.

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100