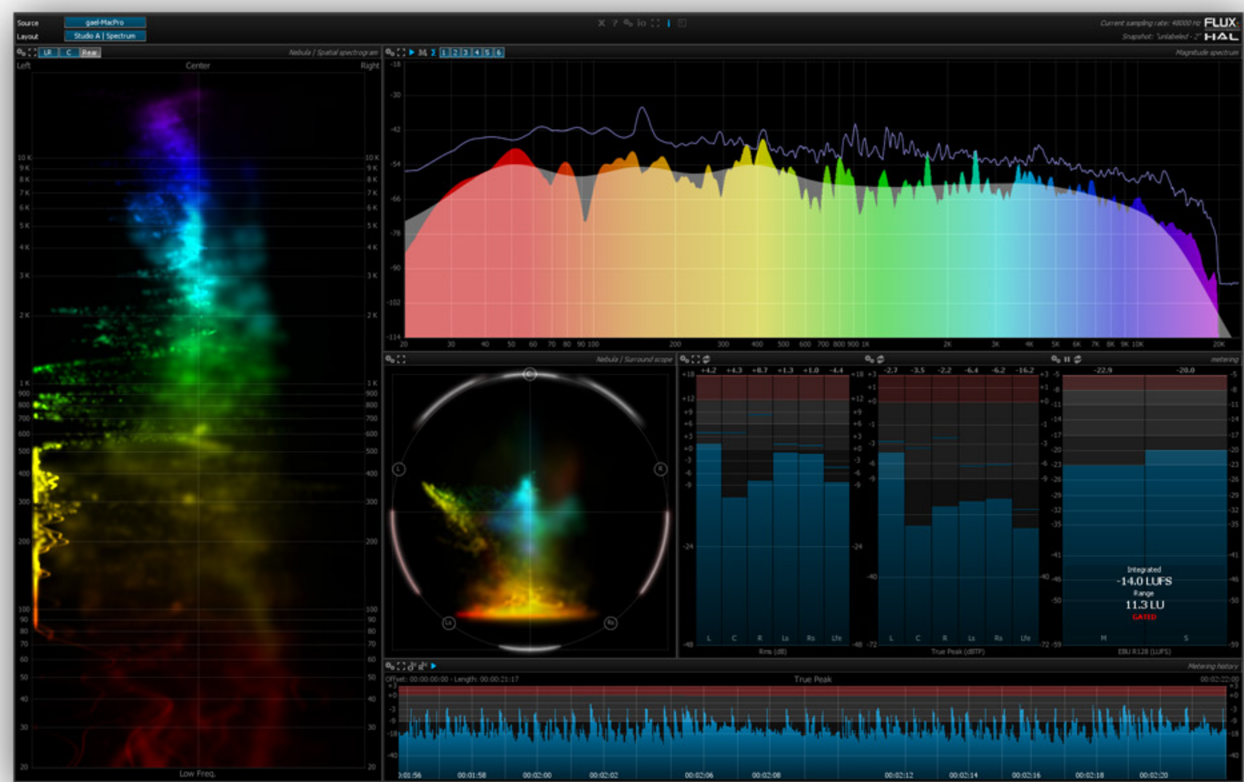
Pure Analyzer System
User manual



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1 Welcome

Thank You!

Thank you for purchasing Flux:: Pure Analyzer System.

1.1 Pure Analyzer Versions

The Pure Analyzer is available in two different versions, The Pure Analyzer Essential, and the Pure Analyzer Studio Session.

The main difference between the two are:

	Pure Analyzer Studio Session	Pure Analyzer Essential
Inputs/Outputs	Mono / Stereo	Mono / Stereo MultiChannel* up to 16 ch.
I/O Configuration	SampleGrabber Plug-in (No Hardware I/O)	SampleGrabber Plug-in Hardware I/O: ASIO / Core Audio
Sample Rates (kHz)	44,1, 48, 88,2, 96	44,1, 48, 88,2, 96 176,4, 192, 384 DXD
Supported Options		Live / Metering MultiChannel

*Multichannel operation requires the Multichannel add-on option.

2 Initial setup

2.1 Introduction

In a conventional digital system, audio material is captured, stored, transmitted and reproduced as a sequence of values, which correspond to the amplitude variations of an electric signal at discrete points in time. Our ability to extract meaningful information from this raw data through either hearing or visualization of the signal curve is however somewhat limited to emotional interpretation, which as one may expect, is extremely subjective.

Extensive studies have shown that first converting this data to a so-called frequency representation is extremely useful for a broad range of audio applications, as it is quite similar in principle to the human auditory system. A proper detailed explanation of the reasons behind this is well outside of the scope of this manual, so we will only hint at a few important characteristics of human hearing, namely its

- ability to recognize and isolate sounds base on their relative intensity or loudness
- ability to identify a pitch and timbre (color, texture) for sounds that fall in this category
- ability to distinguish sounds based on their actual or perceived location

A number of methods have been invented in order to translate these characteristics to measurable quantities that can be expressed in standardized units. These provide invaluable tools for assessing the quality of an audio recording, assisting the engineer in detecting potential mix problems, conforming to industry standards and requirements, calibrating loudspeaker systems, tuning room acoustics, etc.

A fundamental tool for transforming a time-based digital audio signal into a frequency-based representation, a.k.a frequency spectrum, is the discrete Fourier transform (DFT) and its derivatives, such as the Short-Term Fourier Transform (STFT) and Fast Fourier Transform (FFT). Basically, the DFT maps a signal to a set of amplitudes taken at equally-spaced frequency intervals. In essence, one can see the DFT as a bank of many band-pass filters, with as many meters at the output of these filters.

Whilst constraining the frequencies to be taken at fixed, regular intervals, is convenient both in terms of processing resources and simplicity of the computation, amongst other reasons, this linear frequency binning does not represent human hearing, which is essentially logarithmic (constant Q), very accurately. The analysis engine in Pure Analyzer therefore offers both options, which are discussed in more detail in Spectrum analyzer (see p.27).

2.2 Samplegrabber

2.2.1 Principle of operation

Pure Analyzer System completely separates signal acquisition from analysis for maximum flexibility.

Source and response signals are first acquired by the SampleGrabber plugin, and subsequently routed across the network using the ZeroConf/Apple Bonjour protocol. Finally, the Pure Analyzer standalone applications receives the sample feed(s) and analyzes them.

SampleGrabber is a surround-capable plugin, available in all common formats (VST, AU, RTAS and TDM), whose channel configuration is set automatically, or by clicking the icon and setting the desired channel count in the I/O sub-menu.

The Pure Analyzer application displays a list of SampleGrabber instances found on the network in the Audio source (see p.10) menu. Each instance is identified by the associated computer network name it is running on. Clicking a name in the list selects the corresponding SampleGrabber for input.

Notes

You can insert up to 64 instances of SampleGrabber plugins inside one same DAW, and up to 64 Pure Analyzer instances can be connected to any SampleGrabber instance over the network. A SampleGrabber can be connected to up to 64 Pure Analyzer instances over the network.

We do however recommend to limit the number of instances in order to avoid saturating the network.

2.2.2 Network configuration

Network configuration is completely automatic and transparent, thanks to the use of the ZeroConf/Apple Bonjour protocol. Should you encounter any problems with your connection, we advise you to check wether the UDP port range from 46000 to 46064 is opened in your firewall, for both incoming and outgoing connections.

The audio transport requires approximately 1.4Mbps for each channel at a sample rate of 44.1kHz, whereas a 5.1 configuration at 96kHz demands a little less than 20Mbps. A properly functioning standard Ethernet 100Mbps network should therefore be more than sufficient to handle most scenarios.

Notes

The above bandwidth requirements naturally do not apply when using both SampleGrabber and Pure Analyzer on the same machine.

Please check with your network administrator if you have any bandwidth issues and/or special requirements.

2.2.3 Password

An optional password, which is a simple 4 digit number, allows you to apply light encryption to the audio stream for secure transmission over the network. It is set to 0 by default which disables encryption; in this case no additional action in the Pure Analyzer application is required on your part.

If you wish to employ and define a password in SampleGrabber, please enter a matching value in the SampleGrabber (see p.13) menu of the Pure Analyzer application in order to be able to decrypt the incoming stream.

Please note that the security level provided by this encryption is mild, and is only intended to protect from anyone eavesdropping your audio stream inside the internal network. It is not intended as a substitute for conventional network security practices and measures such as software and hardware firewalling, etc.

2.3 Typical configurations

2.3.1 Autonomous mobile configuration

Capture

- Required:
  - Entry-level laptop.
  - Sound card with at least one stereo input available.
  - Basic DAW software for SampleGrabber signal capture, capable of using of one of the supported plugin formats.
  - Network connection.
- Optional:
  - Phantom-powered microphone input for capturing room signal in performing room acoustics measurements (Transfer function and impulse response).
  - Wireless network connection.

IMPORTANT! - The Pure Analyzer Studio Session supports audio input only by using the SampleGrabber plug-in. No hardware input/output options are supported for the Pure Analyzer Session.

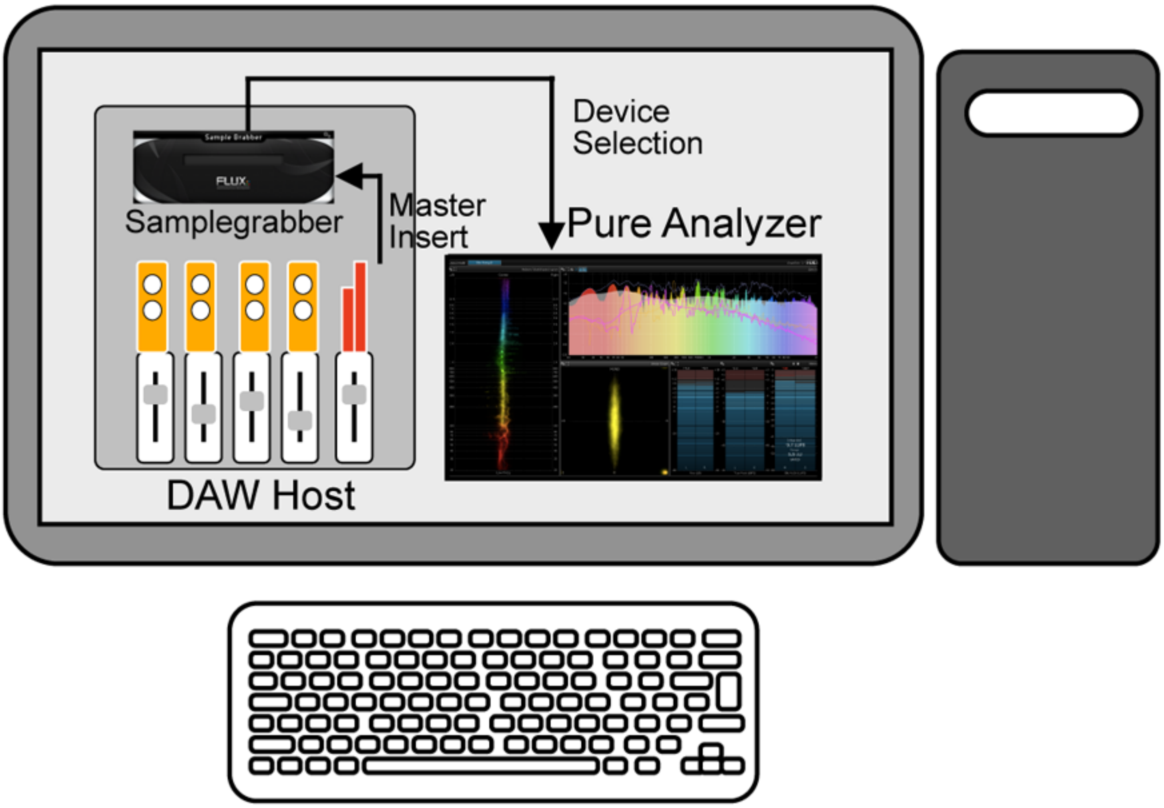
Analysis

Mid-range desktop with an OpenGL/DirectX capable graphics card that meets the minimum System requirements (see p.112)

Capture and analysis can naturally be performed on the same machine, although you can also couple the system with a wireless transmitter to route a source test signal to the laptop in order to perform transfer curve measurements at different locations more conveniently.

2.3.2 Digital Audio Workstation

Any recent computer should be able to run Pure Analyzer smoothly in a stand-alone configuration. Running your preferred DAW host application alongside with an instance of Pure Analyzer naturally raises the requirements. Operating in this way will most probably require a dual-screen setup in order to be able to monitor the Pure Analyzer display and DAW interface simultaneously.

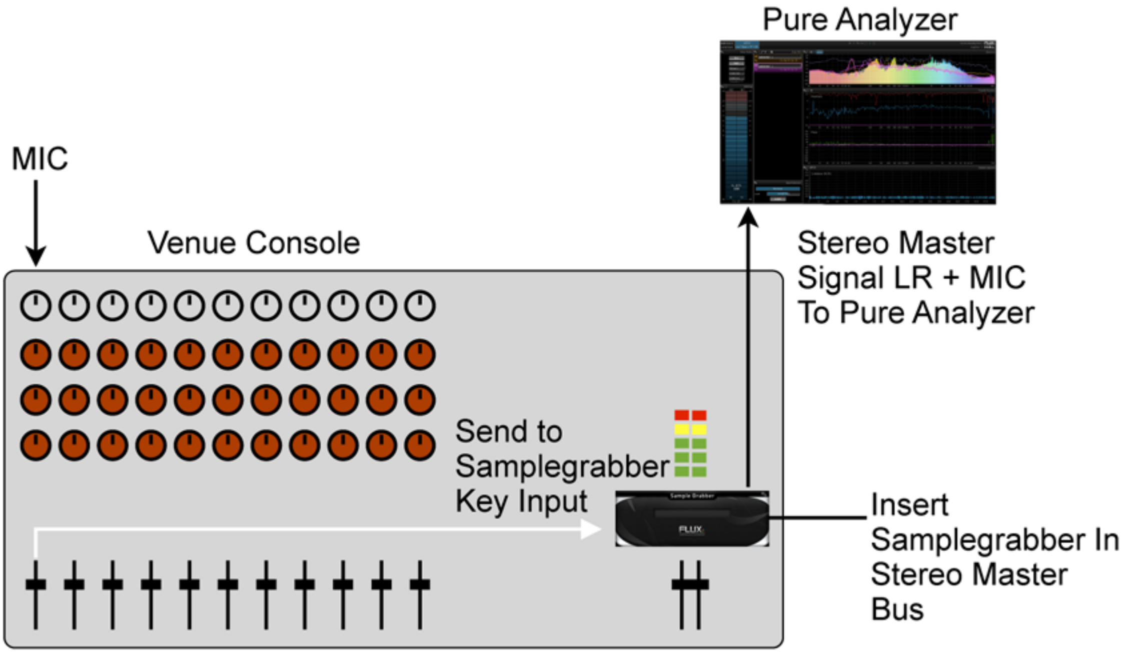


SampleGrabber and Pure Analyzer running on the same machine

2.3.3 Avid Venue Console

SampleGrabber is available as a AAX DSP plugin, which is the preferred format when using an AVID Venue live console. Using one or more SampleGrabber instances will free up several precious hardware outputs.

When performing transfer function and impulse response measurements, a recommended way of working is to insert a SampleGrabber on the master output and set the microphone signal as key input. This simplifies the routing and allows for fast switching between different microphones.



Recommended setup with Avid Venue console

3 User interface

3.1 Common workflow

3.1.1 Mouse commands and conventions

The following mouse click actions are available:

Left-click

Selects the active element.

Right-click

Toggles the display of the corresponding setup menu for the item beneath the current mouse location.

Modifier + click

Ctrl-click is equivalent to right-click.

Inside a setup menu item, Alt-click resets the corresponding setting to its default value.

Alt-clicking inside an item, with a zoom factor greater than one, reset the current zoom to full view (Factor = 1).

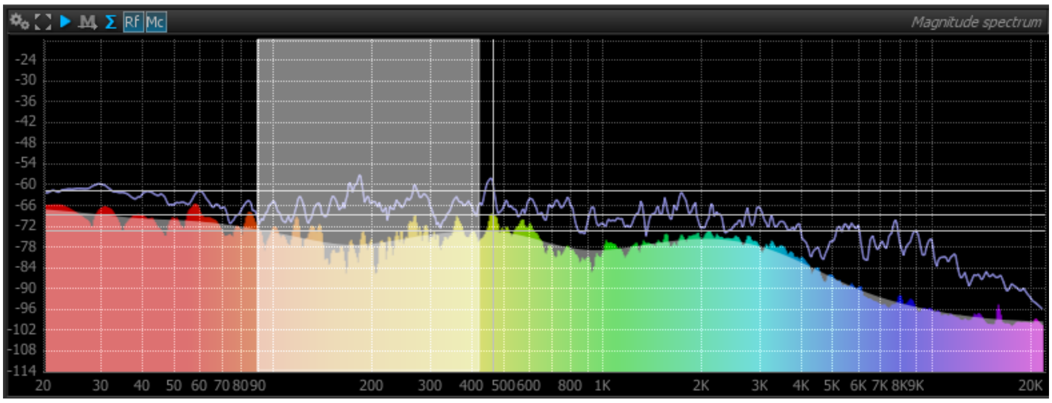
Double-click

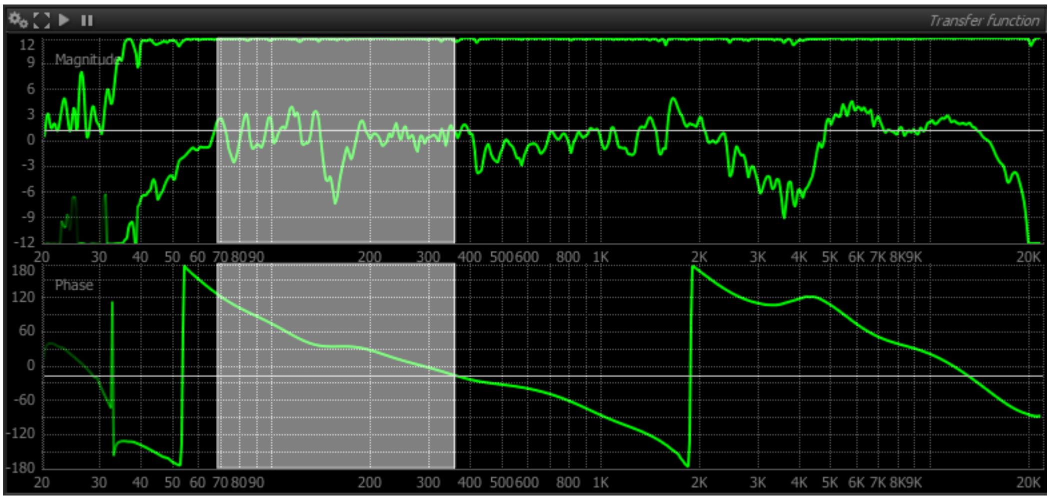
Double-clicking on an editable control such as a slider or text box enters keyboard entry mode, double-clicking again validates the new value. Double-clicking anywhere inside a panel switches the panel to full-window mode, where the whole application screen is occupied by the corresponding panel; double-clicking a second time reverts to the normal layout.

Click and drag

Clicking + dragging inside an item, with a zoom factor greater than one, shifts the current scale.

Alt + Clicking + dragging inside an item with a Zoom Factor, allow to setup a new zoom according to the defined selection.





Scroll wheel + click and drag

Turning the middle mouse wheel, if present, affects the current horizontal zoom level of the item under the cursor.

Activating the wheel with the middle button simultaneously engaged shifts the current scale when the current zoom factor is greater than one.

3.1.2 Keyboard shortcuts

Main

Toggle full screen mode	*Alt + Return*
Display context help / credits page	*F1*
Reconnect network	*F5*
Switch to next layout	*TAB*
Switch to previous layout	*Shift + TAB*
Toggle mouse info update on/off	*F6*
Toggle real-time curves display	*Enter / Return*

Layout

CTrl + F... Key	*go to specified layout*

Snapshot

Create new snapshot	*Space*
Create new sweep snapshot	*Shift + Space*
Create new average snapshot	*Windows: Ctrl + Shift + Space* *Mac OS: alt + Space*
Update first selected snapshot	*Windows: Ctrl + Space* *Mac Os: Ctrl + Shift + Space*
Delete selected snapshot(s)	*Delete*
Load snapshot project	*Ctrl + O*

Export selected snapshot(s)	*Ctrl + S*
Select all snapshots	*Ctrl + A*
De-select all snapshots	*Escape*
Select next snapshot	*Down Arrow*
Select previous snapshot	*Up Arrow*
Add next snapshot to selection	*Shift + Down*
Add previous snapshot to selection	*Shift + Up*
Select first snapshot	*Home*
Select last snapshot	*End*
Toggle selected snapshot Main curve on/off	*0*
Toggle selected snapshot Coherence curve on/off	*1*
Toggle selected snapshot Mag curve on/off	*2*
Toggle selected snapshot Phase curve on/off	*3*
Toggle selected snapshot Spectrum curve on/off	*4*
Toggle selected snapshot IR curve on/off	*5*

Impulse Response

Add delay	*Ctrl + Add / NUMPAD +*
Subtract delay	*Ctrl + Subtract (NUMPAD -)*

Delay Finder

Increment delay by one sample	*Add (NUMPAD +)*
Decrement delay by one sample	*Subtract (NUMPAD -)*
Find delay	*Ctrl + F*
Reset delay	*Ctrl + NUMPAD 0*
Compensate delay	*Ctrl + D*

Generator

Toggle generator on/off	*G*

Meters

Refresh all meters	*M*

Metering history

Set TimeCode offset	*T*
Reset TimeCode offset	*R*

3.2 Audio source

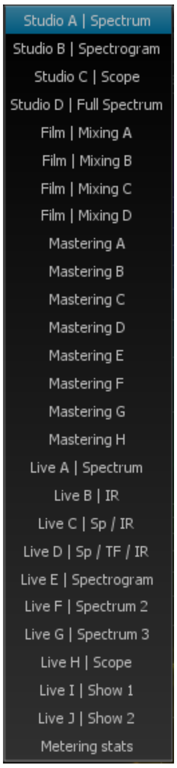
Audio source allows you to select which source to use as input. Depending on your current configuration and settings, this will include:

- Available SampleGrabber instance(s), either local or remote.
- Available hardware device(s), if one or several sound cards are present on the host system, and the corresponding device has been selected in the Hardware IO (see p.21) configuration dialog.

IMPORTANT! - The Pure Analyzer Studio Session supports audio input only by using the SampleGrabber plug-in. No hardware input/output options are supported for the Pure Analyzer Session.

3.3 Layout mode

Pure Analyzer offers a number of user interface layouts designed and named according to typical tasks:



The layouts are grouped into categories, as described below.

Studio

For recording and mastering studio applications, these layouts allow simultaneous monitoring of the spectrum amplitude and spatial distribution, program level and phase.

Film mixing

Provide an overview of the signal amplitude spectrum, phase and and levels.

Film C & D provide Stereo Vector Scope + phase in addition.

Mastering

Special emphasis is put on controlling program level, spectral equilibrium and spatial image. These layouts all offer a Nebula | Spatial spectrogram, a Vector/Surround Scope, Spectrum Amplitude and Level Meters, in different size combinations.

Live sound system alignment

These layouts provides the elements needed by the live sound engineer when performing speaker array calibration tasks, namely delay finder, level meter, transfer function magnitude, phase and coherence spectra, impulse response, and snapshot facilities.

Live - Show

These layouts are intended for use by a live sound engineer during the course of a show, allowing for constant monitoring of the principal level and spectral indicators of the FOH mix.

Metering statistics

Overview of all metering data at a glance.

<table>
<tr><th>Layout</th><th>Nebula | Spatial Spectrogram</th><th>Nebula | Surround Scope</th><th>Vector Scope</th><th>Magnitude Spectrum</th><th>Spectrogram</th><th>Metering</th><th>Metering History</th><th>Metering Statistics</th><th>Wave Scope</th><th>Transfer Function</th><th>Impulse Response</th><th>Live IO</th><th>Signal Generator</th><th>Snapshot</th><th>Leq</th></tr>
<tr><td>Studio A | Spectrum</td><td>●</td><td colspan="2">●</td><td>●</td><td></td><td>●</td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Studio B | Spectrogram</td><td>●</td><td colspan="2">●</td><td></td><td>●</td><td>●</td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Studio C | Scope</td><td>●</td><td>●</td><td>●</td><td></td><td></td><td>●</td><td>●</td><td></td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Studio D | Full Spectrum</td><td>●</td><td>●</td><td>●</td><td>●</td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Film | Mixing A</td><td></td><td colspan="2">●</td><td>●</td><td></td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Film | Mixing B</td><td>●</td><td colspan="2">●</td><td>●</td><td></td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
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<tr><td>Mastering B</td><td></td><td colspan="2">●</td><td>●</td><td></td><td>●</td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Mastering C</td><td>●</td><td colspan="2">●</td><td>●</td><td></td><td>●</td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Mastering D</td><td>●</td><td colspan="2">●</td><td>●</td><td></td><td>●</td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Mastering E</td><td></td><td colspan="2">●</td><td></td><td>●</td><td>●</td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Mastering F</td><td>●</td><td colspan="2">●</td><td></td><td>●</td><td></td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Mastering G</td><td>●</td><td colspan="2">●</td><td>●</td><td>●</td><td></td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
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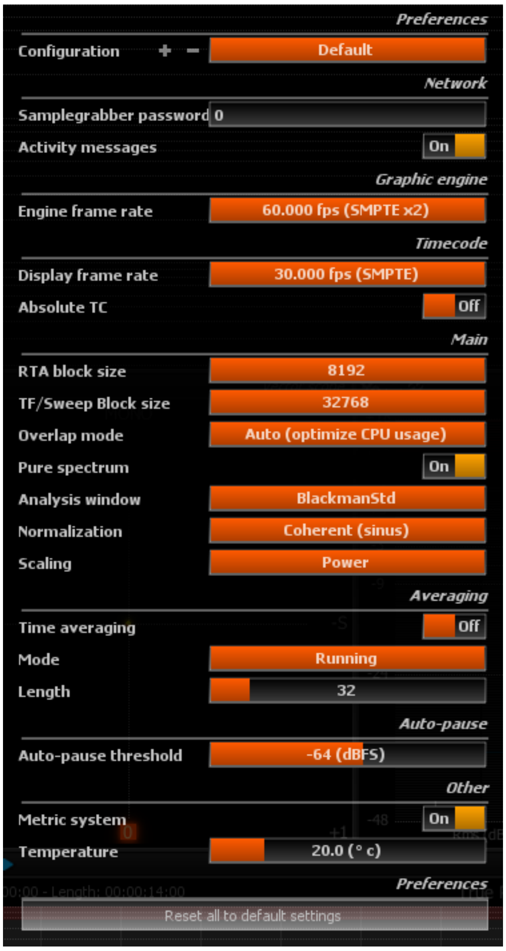
<tr><td>Live A | Spectrum</td><td></td><td></td><td></td><td>●</td><td></td><td>●</td><td></td><td></td><td></td><td>●</td><td></td><td>●</td><td>●</td><td>●</td><td></td></tr>
<tr><td>Live B | IR</td><td></td><td></td><td></td><td></td><td></td><td>●</td><td></td><td></td><td></td><td>●</td><td>●</td><td>●</td><td>●</td><td>●</td><td></td></tr>
<tr><td>Live C | Sp / IR</td><td></td><td></td><td></td><td>●</td><td></td><td>●</td><td></td><td></td><td></td><td></td><td>●</td><td>●</td><td>●</td><td>●</td><td></td></tr>
<tr><td>Live D | Sp / TF / IR</td><td></td><td></td><td></td><td>●</td><td></td><td>●</td><td></td><td></td><td></td><td>●</td><td>●</td><td>●</td><td>●</td><td>●</td><td></td></tr>
<tr><td>Live E | Spectrogram</td><td></td><td></td><td></td><td></td><td>●</td><td>●</td><td></td><td></td><td></td><td>●</td><td></td><td>●</td><td>●</td><td>●</td><td></td></tr>
<tr><td>Live F | Spectrum 2</td><td></td><td></td><td></td><td>●</td><td>●</td><td>●</td><td></td><td></td><td></td><td></td><td></td><td>●</td><td>●</td><td>●</td><td></td></tr>
<tr><td>Live G | Spectrum 3</td><td></td><td></td><td></td><td>●</td><td>●</td><td>●</td><td></td><td></td><td></td><td></td><td></td><td>●</td><td>●</td><td>●</td><td></td></tr>
<tr><td>Live H | Scope</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td>●</td><td>●</td><td></td><td>●</td><td>●</td><td>●</td><td></td></tr>
<tr><td>Live I | Show 1</td><td>●</td><td></td><td>●</td><td>●</td><td></td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td>●</td></tr>
<tr><td>Live I | Show 2</td><td>●</td><td></td><td>●</td><td>●</td><td>●</td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td>●</td></tr>
<tr><td>Metering Stats</td><td></td><td></td><td></td><td></td><td></td><td></td><td>●</td><td>●</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
</table>

Layout contents matrix

Notes

Some layouts might not be available in your Pure Analyzer edition.

3.4 Main setup



Main setup dialog

Configuration

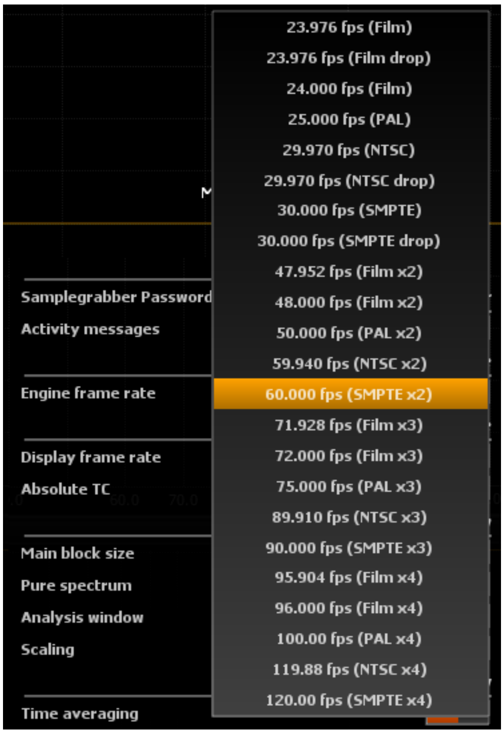
Save / restore a user-defined configuration to and from disk, including all the settings in this panel, as well as IO Configuration (see p.21) and UI Setup (see p.18).

3.4.1 SampleGrabber

SampleGrabber password

The password entered in this field should match the one used by the SampleGrabber you wish to use as a source. This provides a reasonable level a security and prevents unauthorized access to your audio material broadcast over the network. Please take into consideration the encryption used only provides moderate protection, and is not intended to replace other security guards such as firewalls etc.

3.4.2 Graphic engine



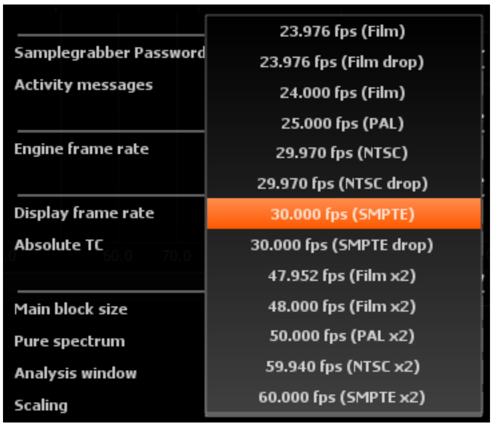
Available graphic engine frame rates

Here you can specify the rate at which the display should be refreshed. Please note higher frame rates place higher demands on the GPU, and to a lesser extent, on the CPU.

Notes

The effective frame rate can be displayed by typing `SetRenderStats(1)` in the console.

3.4.3 Time code



Available display frame rates

Display frame rate

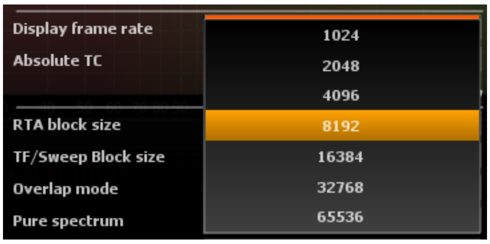
Sets the frame-rate used for time display in various parts of the program. Set it to match the frame-rate of your source material to facilitate locating time events, when working with film, TV or other time-stamped material.

Absolute TimeCode

This settings toggles between absolute and relative time-code display formats. Absolute TimeCode is taken from the time the application was started. Relative TimeCode is the time-elapsed since the TimeCode offset position. See metering history usage (see p.71) for information on working with TimeCode.

3.4.4 Main

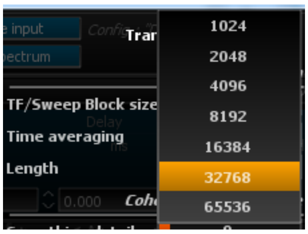
RTA block size



Defines the size of the blocks, in samples, fed to the main spectrum analyzer engine, which is used by the spectrum magnitude, Nebula and spectrogram views.

Pure spectrum Toggles between optimized frequency analysis (default) and standard FFT.

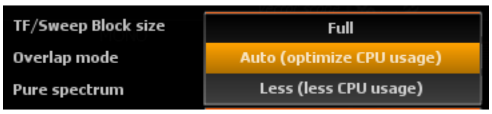
TF/Sweep Block size



Block size used for the transfer function and snapshot performed with sine-sweep. The default is 32768, which is appropriate for most cases.

Increasing this value gives better frequency resolution, at the expense of CPU load. Lower values can be employed if you're only interested in the overall response of the analyzed system.

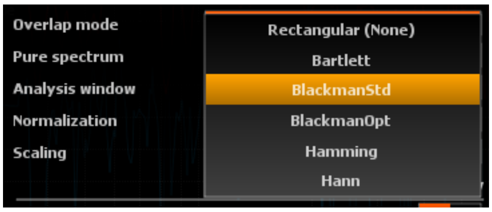
Overlap Mode



The overlap mode setting determines how much incoming audio frames overlap each-other. A higher overlap results in a smoother display update, at the expense of increased CPU usage. The available settings are

- Full: highest overlap
- Auto: optimizes overlap depending on available CPU resources (Default).
- Less: minimizes overlap for minimal CPU usage (useful for slow machines)

Analysis window



Selects the analysis window applied to the incoming blocks.

Available choices are:

- Rectangular (None).
- Bartlett.
- Blackmann standard (default).
- Blackmann optimized.
- Hamming.
- Hann.

There is no reason to change this setting unless you have a specific reason to do so and fully understand the implications.

Normalization

Selects the normalization mode used to normalize the global gain of the spectrum display.

Available choices are:

- Coherent (sinus). 0dB peak sine gives 0dB amplitude
- Incoherent (noise/music). 0dB RMS noise or music gives 0dB power

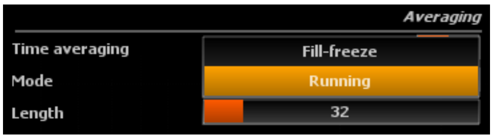
Scaling

This setting controls the frequency dependent amplitude spectrum correction curve.

Available choices are:

- Amplitude: equivalent to no scaling. Amplitude of pure tones at different frequencies register at the same value. Incoming white noise is displayed as a (quasi) flat curve.
- Power (default): scaling inversely proportional to frequency (1/f). Incoming pink noise is displayed as a flat curve.

3.4.5 Averaging



Time averaging

Engages averaging of spectrum magnitudes over time. Default is off.

Mode

- Running: the average display is updated as soon as a new incoming block arrives. This is the default.
- Fill-freeze: the display is only updated when a fresh batch of N new incoming blocks has arrived. The display is frozen until the next batch of N blocks arrive, and so on. N corresponds to the length setting defined below.

Length

The number of incoming blocks over which the resulting average spectrum is computed. Lower values lead to faster apparent display update rates, while higher values smooth-out any time-variations more. Default is 32.

Remarks

Running average employs a weighting window that gives more importance to the last incoming blocks of samples. This type of time averaging is also called *moving average*, *rolling average* or *running average*, and is good for smoothing out abrupt variations in time and still be able to monitor in a continuous fashion.

Fill-freeze mode is useful for stabilizing a flickering display while still following long-term variations, which permits a more detailed study of the curve(s). This mode is therefore useful to get a very steady picture of the spectrum while still monitoring some of the mid-term changes, and saves you from holding and resetting the display manually again and again.

3.4.6 Various

Auto-pause threshold

Analysis is paused whenever the level of any channel of the incoming audio falls below this level. Set this a tad above the acoustic and electronic noise floor of your input signal chain to retain measurements even though the audio (music program or test signal) has stopped.

Metric system

Toggle displayed units between:

- Metric system (default): distance expressed in meters, temperature in degrees Celsius.
- Imperial units: distance expressed in inches and feet, temperature in degrees Fahrenheit.

Temperature

This should be set to the ambient temperature at the current location in order to get the most accurate time to distance conversions in the delay finder and impulse response panels. The following table gives an idea of how much the speed of sound varies with temperature.

Temperature (°C)	Speed of sound (m/s)
0	331.3
15	340.31
25	346.18
35	351.96

Preferences reset

Resets "Default" application configuration settings to their default initial value. Please note the changes are only effective after restarting the application.

3.5 UI Setup

User interface setup dialog

Configuration

Saves / restores a complete user defined configuration.

Fonts: Small Scale

Sets the size of the smallest font used for drawing the grid labels.

Fonts: Large Scale

Sets the size of the largest font used for drawing the grid labels.

Fonts: Spectrum Peak Label

Sets the size of the font used for the Spectrum peak label (see p.34).

Brightness

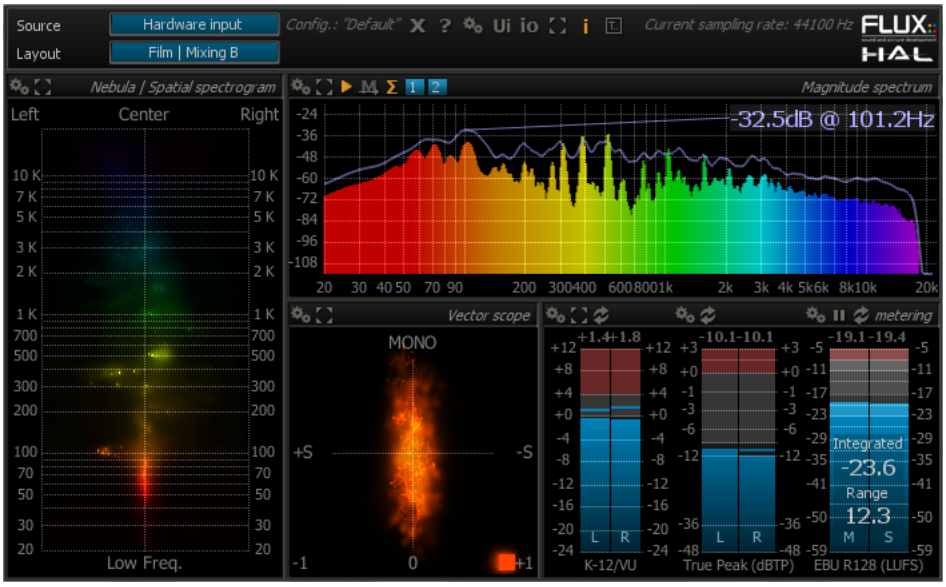
Adjusts global user interface brightness.

Contrast

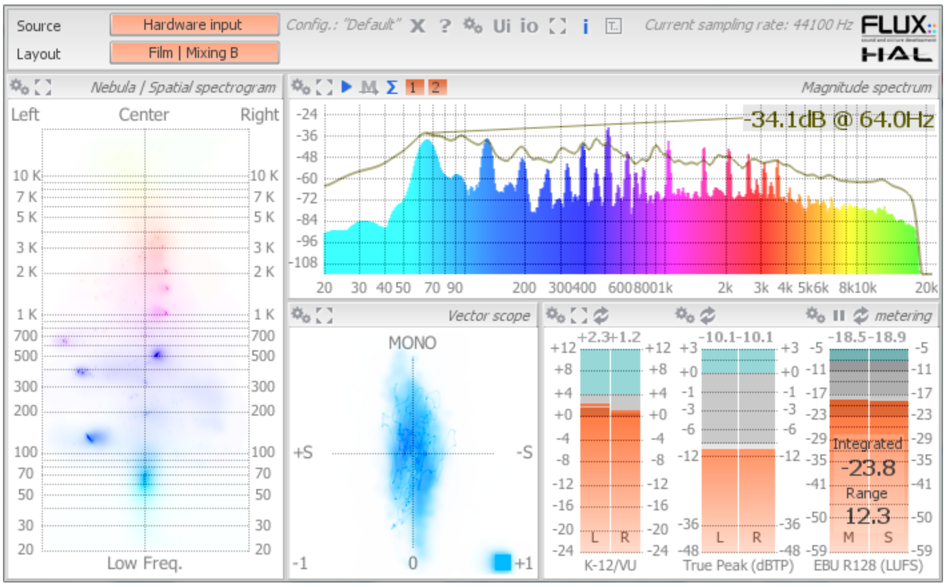
Adjusts global user interface contrast.

Reverse color scheme

When engaged, the user interface color scheme switches from white/grey on black to black/grey on white, for improved readability in an outdoor environment.



Reverse color scheme off

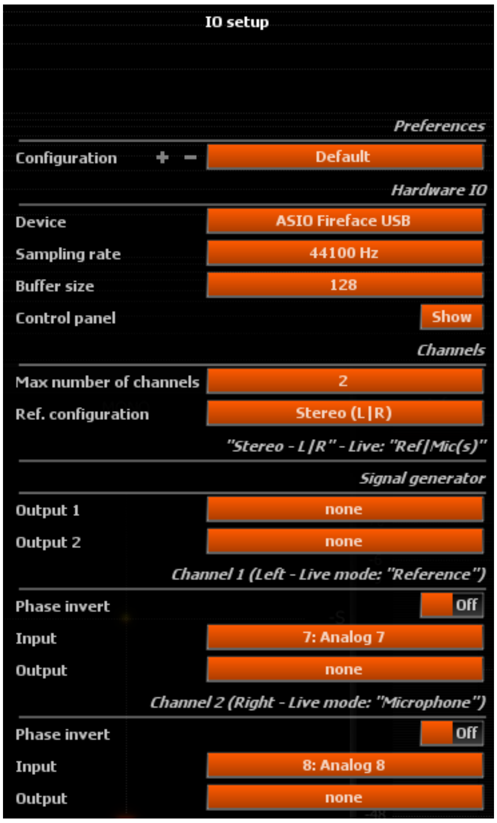


Reverse color scheme on

Layout Shortcuts

This list allows you to set up to nine shortcuts for direct access to your most frequently used layouts.

3.6 IO Configuration



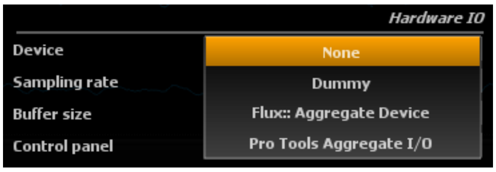
IO configuration dialog

Configuration

Saves / restores a complete user defined configuration.

3.6.1 Hardware IO

Device



This setting lets you choose amongst a selection of devices, depending on your particular hardware configuration.

IMPORTANT! - The Pure Analyzer Studio Session supports audio input only by using the SampleGrabber plug-in. No hardware input/output options are supported for the Pure Analyzer Session.

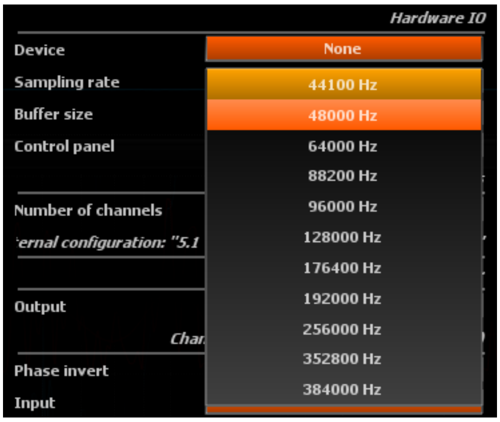
None

This disables hardware input and output altogether. This is the recommended choice if you do not want to take advantage of Pure Analyzer's built-in audio capabilities, for example if you're working with a SampleGrabber inside a DAW or Avid Venue console setup. With some sound cards that aren't multi-client capable - meaning only one program can access it at once - disabling I/O is necessary to continue using another program simultaneously.

Your soundcard

Any installed soundcard(s) will be listed here. Under Windows, it might appear several times, in which case be sure to select the native ASIO driver for performance, not an emulated driver which be labeled something like ASIO DirectX Full Duplex Driver, Generic Low Latency ASIO Driver or similar.

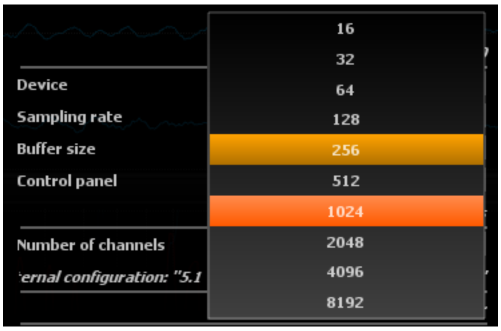
Sampling rate



Available sampling rates (hardware specific)

Sets the sampling rate used internally by the application. When a hardware device is selected, be sure to match this to the sampling rate set in the application panel of your soundcard control panel. We deliberately chose not to employ resampling, which in our opinion has no place in a measurement instrument. Instead we generally advise you to set your soundcard's sampling rate to 44.1k or 48k, which covers the entire audio hearing range (20-20kHz). Increasing the sampling rate above these values increases the processing power require to carry out the computations without any benefit for most practical applications.

Buffer size



Displays the current soundcard I/O buffer size. Depending on your soundcard, you might be able to change this to a different value directly in Pure Analyzer without opening its control panel beforehand. Smaller buffer sizes leads to a shorter latency between incoming audio, display update, and audio output. This setting is certainly not as crucial as in the context of live sound processing, so there is no need to go down to extremely small values here, as this only increases the system load without offering any practical advantage.

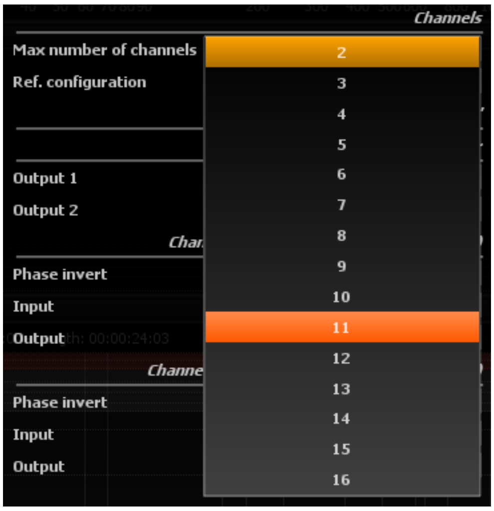
Keep in mind a display refresh rate of 60Hz means one frame lasts for approx. 16ms, which is a bit longer than one 512 buffer at 44.1kHz, so the display will always lag less than one frame after the audio with such a setting.

Control panel

Opens the ASIO (Windows) / CoreAudio (MacOS) control panel for the selected soundcard driver, where you can make further settings depending on your particular hardware, such as routing, input gain etc.

3.6.2 Channels

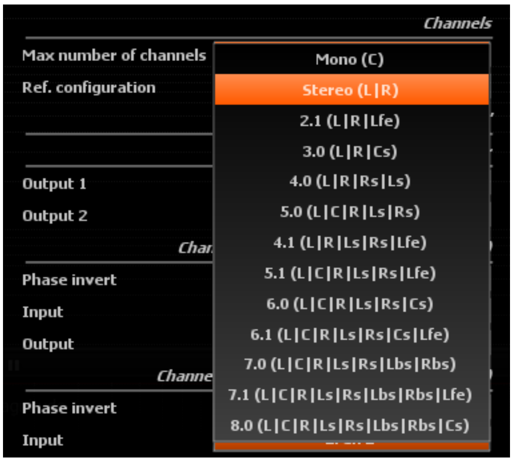
Max number of channels



Selects the maximum number of channels to be used by the application, or equivalently the number of channels in the application I/O buss. You should set this according to the source material format you want to analyze and visualize. This determines notably how many real-time curves are displayed in the Spectrum analyzer (see p.27) view, wether or not the Surround scope (see p.48) is displayed, etc.

IMPORTANT! - The Pure Analyzer Studio Session supports only 2 channels of audio.

Reference configuration



Reference configurations available with 8 max. channels

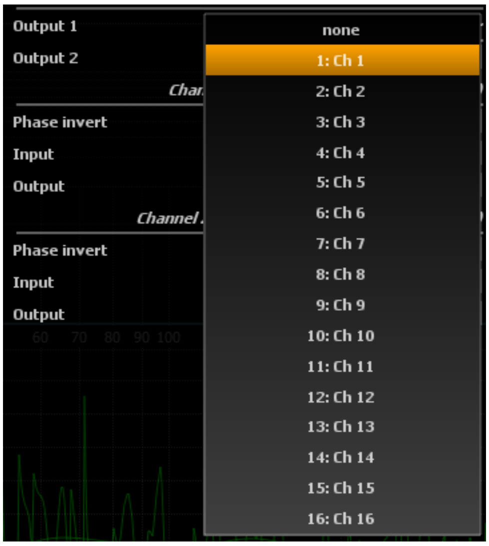
Depending on the setting above, the possible standard channel configurations will be listed here, and will be a subset of the following:

- Mono (C): single center channel
- Stereo (L|R): two left-right channels
- Surround: various standard configurations depending on the exact channel count

The channels are labeled according to this configuration to make them easier to identify.

3.6.3 Signal generator

Output



Example output channel routing (hardware specific)

Selects one or two physical channels to which the Signal generator (see p.87) output should be sent.

Notes

In case of stereo output, the signal is identical on both channels. This is provided as a facility for soundcards with minimal routing capabilities, and to avoid using a Y patch cable.

3.6.4 Channel 1 / Channel 2

The following group of settings are displayed for every channel selected in Channels (see p.23). The heading displays the channel number, followed by its name, and wether it corresponds to the reference or microphone input signal for the first two channels in Live IO (see p.82) mode.

Phase invert

When engaged, the phase of the corresponding channel is inverted in order to compensate for a reverse polarity somewhere else in the signal chain. Default is off.

This can happen with incorrect or non-standard wiring, when a phase switch is engaged on the preamplifier, an analog device has an odd number of inverting stages … Use this with caution as it can compromise measurements if the “real” input signal phase does not match.

Input

Selects which hardware device input should be routed to the corresponding internal application input.

Output

Selects which hardware device output the corresponding internal application output should be routed to.

3.7 Other

3.7.1 Hold info text

When this button is disengaged, textual information overlays displayed above curves are held until the button is engaged again. This allows you to check a particular value precisely, such as an amplitude, gain, or phase at a particular frequency determined by the mouse cursor position when the switch was engaged. The most convenient to use this feature is to use the corresponding keyboard shortcut (*F6*).

3.7.2 Full-screen mode

Toggles full-screen mode on and off, to maximize screen real estate by masking the task bar (Windows) or Dock (MacOS) if desired.

3.7.3 Close

Exits the application.

3.7.4 Help / about

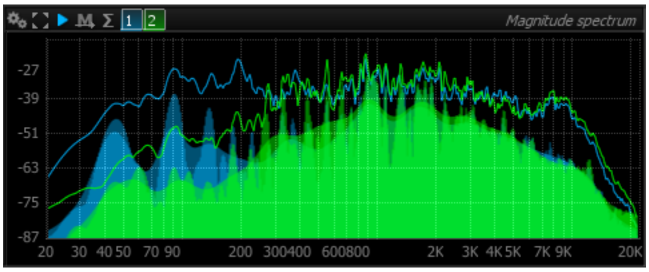
Displays the application credits, Pure Analyzer software version number, the options available with the current license, as well as a table summarizing assigned keyboard shortcuts.

4 Spectrum analyzer

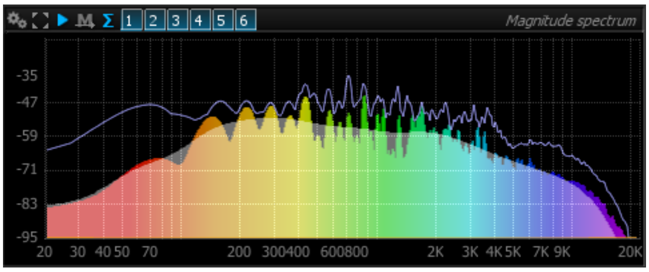
4.1 Basic principles

The global principle and purpose of a spectrum analyzer is to transform an incoming signal, which is basically a series of amplitudes taken at successive points in time, into a series of values versus frequency. Transforming an audio signal onto a frequency scale is indeed of great interest for a wide range of tasks, and notably allows one to display a global, perceptually meaningful and precise picture of the audio contents.

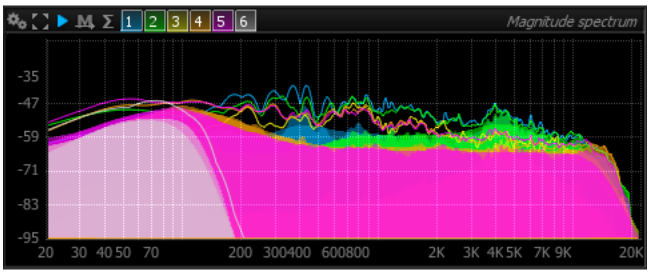
The display represents the so-called magnitude spectrum of the incoming signal, which is a two-dimensional curve of the magnitudes of the signal taken at frequencies ranging from 0 (DC) to half that of the current sampling rate (or Nyquist frequency in signal processing jargon). This is probably the most commonplace and most easily understood spectrum analyzer visualization, and is the place where you should start most of the time when you want to inspect the frequency content of your audio material.



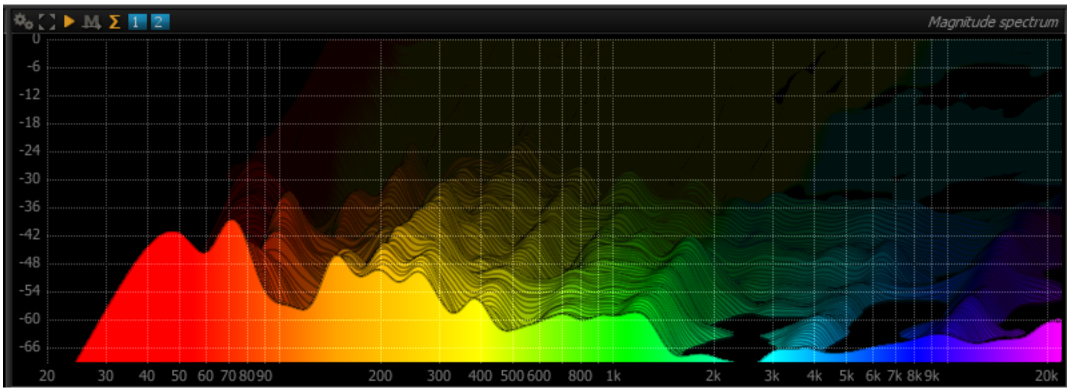
Magnitude spectrum of a stereo signal with summing disabled, max and smoothed curves enabled



Magnitude spectrum of a 5.1 surround signal sum with max and smoothed curves enabled



Magnitude spectrum of a 5.1 surround signal with summing disabled



Magnitude spectrum with Slide option enabled (Real time waterfall)

Magnitude spectrum setup dialog

4.2 Block size

Keep in mind that the incoming audio needs to be accumulated in a buffer for a certain amount of time before the data can be computed and the display updated. In contrast with the buffers you probably know from soundcards, this block-processing is not just a computer technicality and only a source of undesirable latency, but an integral part of the process related to the mathematical aspects involved (Time-frequency product uncertainty principle).

As such it determines both the precision of the analysis and the maximum display rate, and should be adjusted depending on the specifics of your application.

Remarks

The default setting is 8192 samples, which corresponds to a length of roughly 180ms at 44.1kHz sampling rate. This value constitutes a good compromise between precision and responsiveness for most situations. However, if you need to measure a particular frequency with great precision, you should raise the analysis block size. On the other hand, if you need to follow rapid spectrum variations, this value should be lowered.

Notes

In order to maintain a sufficiently responsive display refresh rate, blocks overlap by 75 %.

4.3 Transform type

The discrete Fourier transform (DFT) is the traditional method employed to compute the frequency spectrum of a discrete digital signal. DFT can be seen as a series of notch filters centered around frequency bins that are uniformly distributed along the frequency axis, and of constant width.

The quality factor of a resonant filter, commonly denoted as Q, is defined as the ratio of its bandwidth relative to its center frequency. The DFT process is therefore analogous to a variable Q filter-bank, in other words, its frequency resolution is constant across the spectrum. When applied to sliding blocks, this process is called STFT, for Short-term Fourier transform.

Although convenient in terms of computation, this can be seen as less than ideal for many audio applications, for several reasons, the first and foremost being that human perception of frequency is known to be quasi-*logarithmic*. Logarithmic means that a two-fold increase in frequency translates to a one octave shift, a four-fold increase as a two-octave shift - and not four as this would be the case, were our perception linear in nature.

Pure Analyzer employs both standard DFT and proprietary algorithms that more closely model the human perception. In addition to greatly improving the legibility of the resulting curves, this proprietary transform has the additional benefit of reducing sensitivity to noise in the high-frequency portion of the spectrum especially, and provides more stable readouts.

Remarks

You can of course switch back to standard DFT by disengaging the Pure spectrum button.

4.4 Window type

As previously mentioned, the first step is to split the incoming signal into overlapping blocks. Each block is then multiplied with a so-called window signal prior to the spectrum computation. The purpose of this is to minimize side effects of the block processing, such as introduction of transients at the block boundaries, etc.

The window type to use is set in the Main (see p.15) setup.

Remarks

While the windowing process is implemented in the time-domain, it can be also be seen as a smoothing filter in the frequency domain, and as such the choice of window is a compromise between frequency resolution and immunity to artifacts.

Skipping the windowing process altogether, which is the same as applying a rectangular window, is not recommended. Although the rectangular window provides the best frequency resolution, it has very poor leakage characteristics.

Notes

We suggest you leave this setting to the default unless you are quite knowledgeable with these aspects, or in the case you should need to explicitly recreate a specific measurement such as a particular method specified in a standard's document.

The Wikipedia entry on window functions in the context of signal processing is a good reference if you want to get a more thorough understanding of the subject.

4.5 Ballistics

4

The curve display update speed is controlled by the ballistics settings.

Release time

The release time determines how fast the main curve falls back to zero. Default is 300ms.

Max release time

The controls the release time of the optional *Max* curve, which serves to display the medium-to-long term tendency of the magnitude spectrum. Longer times mean curve maxima/peaks will be seen for a longer period of time.

Default is 50 seconds.

Remarks

The attack time is zero so the curve displays reacts instantaneously to a rising amplitude.

4.6 Averaging

This is a global setting controlled in the Averaging (see p.17) section of the main setup.

4.7 Frequency scaling

Scaling controls how the scaling applied to spectrum magnitudes. This is a global setting accessed through the Main (see p.15) setup panel.

Scaling controls whether frequency-dependent amplitude scaling should be applied. This affects how various standard reference signals register on the display. The default *power* scaling will result in a signal with spectrum components of *constant power* registering as a flat curve, whilst amplitude will have the same effect for components of constant *amplitude* such as pure tones (sine signal).

The table below shows how the curve appearance depending on the type of input signal. 1/f corresponds to a rectilinear slope on the display with both X and Y axis being logarithmic.

Input signal	Sine	White	Pink noise
Power scaling	1/f	1/f	Flat
Amplitude scaling	Flat	Flat	1/f

Remarks

For monitoring a mix, it makes most sense to use *power* scaling, as this is is the way our hearing responds. If you need to measure a room's acoustic response, an outboard unit or a plugin's frequency response, the system magnitude transfer function is best suited for this purpose and scaling has no effect.

The *amplitude* scaling setting should therefore really be employed if you need to measure relative amplitude values, such as those of sine test tones at various frequencies. Also note that plain DFT corresponds to scaling set to *amplitude*.

The power of a time-signal is proportional to the square of its amplitude, or equivalently, its power in dB is double the amplitude. However, in the case of a spectrum, we are measuring the output of a filter-bank, which reacts very much differently depending on the type of input signal, so the simple previous formula doesn't apply anymore.

4.8 Display range

Display range can be switched from a fixed reference interval to one that automatically adjusts to the current range of spectrum magnitude values. The latter is useful as a set and forget setting and works well to display the most vertical detail, at the expense of losing the ability to visually compare the current values to a reference level.

4.8.1 dB Min/ dB Max

Sets the minimum and maximum magnitude to display, in decibels. This is visible the range of the display that is taken into account when auto-range is off.

Default range is -18dB (min) to -114dB (max).

4.8.2 AutoRange

When engaged, auto-range continuously adjusts the display to the current range of the data.

Default is off.

Notes

A slight envelope is applied to the auto-range values in order to improve legibility, avoiding the display to follow every minor change. Peaks are always registered however, as these provide valuable information that should not be missed.

4.9 Mode

4.9.1 Smoothing mode

Switches between Window (the default) and various per-octave smoothing types.

When *Window* type is selected, a sliding window average of adjustable width is applied to the curve, which results in more or less frequency detail being removed, depending on the Smoothing detail (see p.33) setting.

When any of the Octave types are selected, the average of the spectrum over the corresponding ISO bands is displayed, as as series of horizontal bars. The following series are available

- Octave
- 2/3 octave
- 1/2 octave
- 1/3 octave
- 1/6 octave
- 1/12 octave

4.9.2 Smoothing detail

Controls the amount of frequency detail of the smoothed curve, when using window smoothing. The value roughly corresponds to the maximum number of valleys and peaks that can stand out the smoothed curve. A low value lets the global tendency of the amplitude spectrum pass through, while values above 20 or so preserve more detail such as harmonics and sharp equalizer cuts and boosts. Default is 3.

Remarks

This curves acts as a kind of zoom-out control, as it shows the global frequency content of the signal, leaving out details such as harmonic peaks and variations imputable to transient and noise components. Typical uses for this curve is to monitor the global frequency balance of a mix and to visualize the influence of broad equalizer corrections on the mix.

4.9.3 Curve display

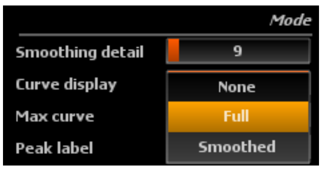
Toggles between the following curve display modes:

- Full: main curve only (no smoothing)
- Smoothed: smoothed curve only
- All: both unsmoothed and smoothed curves

Remarks

Selecting one of the first two modes is recommended to avoid display clutter when comparing several channels and/or snapshots.

4.9.4 Max curve



The max curve employs much longer release time compared to the main curve, and as such registers short peaks much more easily.

The max curve setting controls its visibility and wether smoothing is applied:

- None: curve not displayed
- Full: visible, unsmoothed
- Smoothed: visible, smoothed

Notes

The max curve is never displayed for snapshots, as it would be the same as the main curve, since this type of curve does not evolve in time.

4.9.5 Peak type

This setting controls the manner in which spectrum magnitude peaks are computed:

- Max (global): compute a global maximum over the entire spectrum range.
- Max (user): compute the maximum across a user defined portion of the spectrum set in the Peak range (see p.34).

4.9.6 Peak label

Determines the appearance of the peak display:

- None: peaks are not shown.
- Bar (Full): vertical bar at current peak located at current frequency.
- Bar: vertical bar from base to peak value.
- Mark: text box indicating peak value, in dB, and frequency (Hz) at peak location.
- Mark + Arrow: same as above, with text at the top of the display and arrow pointing at peak location. This is the most precise indication, but takes up more space.

4.9.7 Peak range

Used in combination with the *Max (user)* Peak type setting, this defines the minimum and maximum frequencies to take into account when computing the peak.

4.10 Summation

These settings allow you to modify the appearance of the curves in channel sum mode.

4.10.1 Filled

Toggles wether the main curve is drawn as a a solid-color fill or a plain line.

Default is on.

4.10.2 Width

Thickness of the pen used to draw the curve lines, in pixels.

Default is 1.0.

Notes

This setting also affects individual curves when channel sum mode is disabled.

4.10.3 Full curve color

Color of the pen used to draw the main, full-detail, unsmoothed curve.

4.10.4 Smoothed curve color

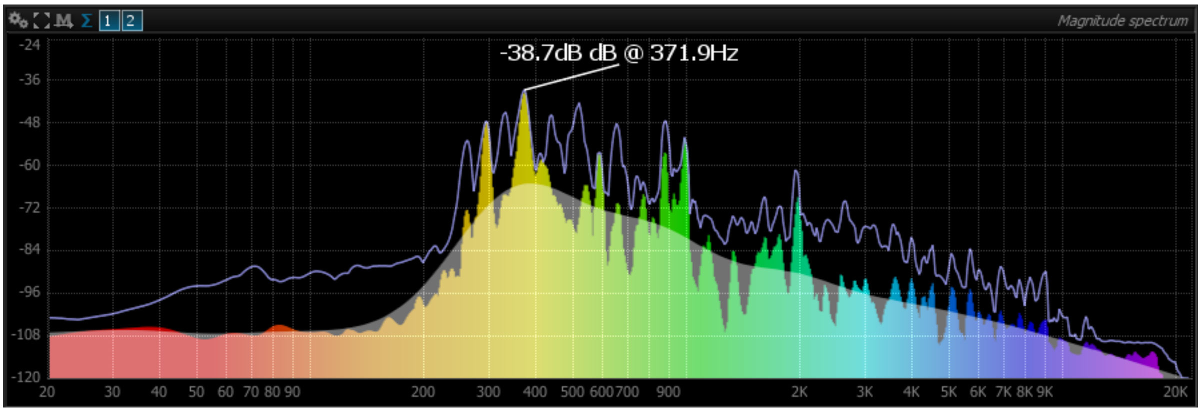
Color of the pen used to draw the smoothed curve.

4.10.5 Max curve color

Color of the pen used to draw the max curve.

4.10.6 Color grading

Applies an optional frequency-dependent coloring to the main channel-sum curve.



Magnitude spectrum with color grading enabled

Remarks

When enabled, any of the above fixed color settings are overridden.

4.11 Channels

This group of settings controls the appearance of curves when channel sum mode is disabled. There is one Ch.*N* curve color setting per channel so you can fine-tune the color scheme employed if you wish to do so.

4.11.1 Filled

Controls wether channel curves are drawn as a solid color fill or a plain line.

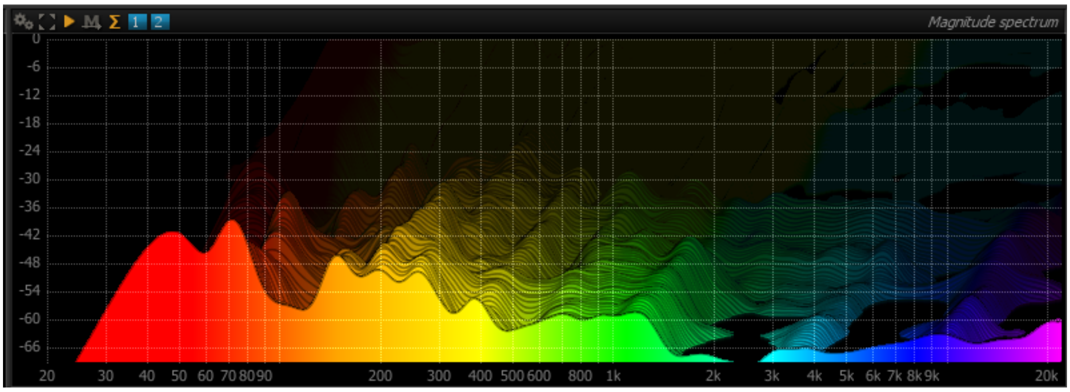
4.11.2 Opacity

Controls the opacity of the fill when *Filled* is enabled. 100% gives a fully opaque fill, lowering this value makes the curve fill more transparent.

4.11.3 Ch.n curve color

This settings controls the color of the curve corresponding to the nth channel, when summation mode is disabled.

4.12 Slide (Real Time waterfall)



4.12.1 Enable

Enable/disable the slide mode

4.12.2 Direction

Define the sliding Direction. From -5 to 5.

Default is 0

4.12.3 Fading

Controls display persistence, *i.e.* the "fade to black" amount for a frame. Lowering this value retains past particles longer, whereas increasing this make them disappear faster.

4.12.4 Blur

Enable / Disable sliding blur

4.12.5 Blur Kernel Size

Controls the radius of the blur effect applied to past particles. Particles are “smeared” more and more as they become older, depending on this setting. Naturally, a bigger value increases the smearing, at the expense of processing power.

Notes

Choosing the value for this setting is really matter of taste, although please keep in mind values that above 5 will require a sufficiently powerful graphics card in order to maintain a responsive display.

4.13 Various

4.13.1 Zoom

This settings allows to check and change the current X-axis zoom level. Default is 1.0, which corresponds to the whole frequency spectrum. Zooming with the mouse is the preferred way, as it offers more control.

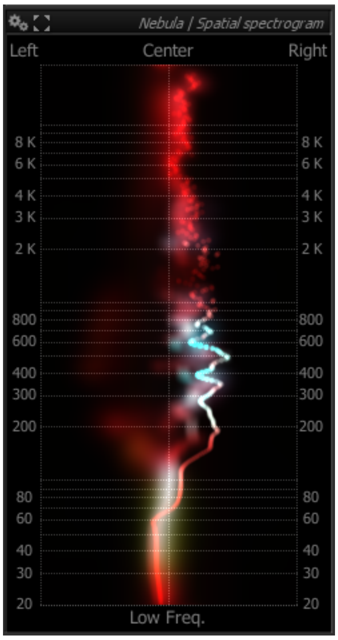
5 Nebula | Spatial spectrogram

5.1 Principle of operation

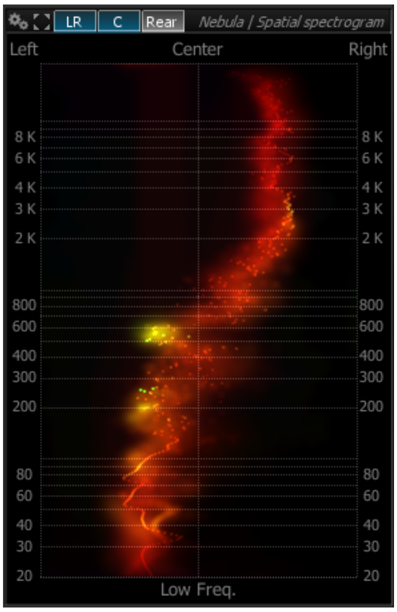
Nebula | Spatial spectrogram provides a unique representation of the audio material in terms of spectral content and localization in the stereo and/or surround space. Its combines the functionality of a spectrum analyzer and a vector scope in a novel real-time display. As such it provides to be an invaluable tool to get a complete and detailed overview of your mix, which you can finely tune in many aspects to suit your particular needs and preferences. A lot of work has gone into optimizing the real-time rendering of the display, not solely for aesthetic reasons, but because we wanted the display to react instantly to all the details in the incoming audio. The idea is literally for you to be able to see what you hear and feel, and not some gross simplification wrapped into shiny eye-candy, however pleasing to the eye.

The overall principles behind Nebula | Spatial spectrogram are quite straightforward:

- At any given time, and for every frequency, the engine computes the position of a frequency in space (2D in stereo , ND for N channel-surround). This position is taken as the center of gravity of the various channels, weighted by the relative amplitude of the signal in their corresponding channel.
- A projection onto a LR-spectrum plane is computed, giving a spectrum-space frame constrained to the stereo field.
- Incoming spectrum-space frames are added back to the previous frames.
- Past frames are progressively "forgotten", using blur and dimming, in order to make place for new information, and increase legibility.



Nebula | Spatial spectrogram display with stereo input



Nebula | Spatial spectrogram display with surround input

Nebula | Spatial spectrogram setup options

5.2 Scale

5.2.1 Focus

Controls the stereo image width X-axis display range, in dB.

A value comprised between ±18 and ±24dB correlates well with our abilities in perceiving the stereo image.

Default is ±18dB.

Remarks

Pixels outside the focus range are clamped to the view boundaries.

5.2.2 AutoScale

This parameter controls whether the intensity of the particles are modulated by the overall audio level variations. In essence, when enabled, the color nuances will vary according to the relative amplitude of a frequency, allowing to monitor the relative amplitude spectrum variations. When disabled, the color will reflect the absolute audio level. You can also think of this as a kind of auto-gain setting.

5.2.3 AutoScale release

This controls whether color variations should be smoothed in time or not. When engaged, color variations is slowed down a bit, which makes overall level transitions more obvious.

Notes

You should to enable this setting when you want to visualize quick level variations such as those that frequently occur in movie soundtracks.

5.2.4 Linear blend range

Adds a constant blend amount to the particle. This ensures some of the particle is always blended into the image even if its original magnitude is low.

A low value for this setting has the effect of stabilizing the appearance of particles. With large values more of the spectrum dynamics are taken into account, and only peaks mostly come through.

5.2.5 Log blending

Toggles between linear and logarithmic blending of the current particle with old particles.

The default is off, ie. linear blending, which tends to favor the display of peaks.

Logarithmic blending on the other hand preserves more of the full dynamic range of the data, and gives some visibility to lower levels also.

5.3 Display

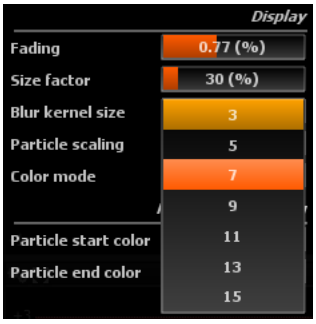
5.3.1 Fading

Controls display persistence, *i.e.* the "fade to black" amount for a frame. Lowering this value retains past particles longer, whereas increasing this make them disappear faster.

5.3.2 Size factor

Controls the size of individual particles with respect to screen size.

5.3.3 Blur kernel size



Controls the radius of the blur effect applied to past particles. Particles are "smeared" more and more as they become older, depending on this setting. Naturally, a bigger value increases the smearing, at the expense of processing power.

Note

Choosing the value for this setting is really matter of taste, although please keep in mind values that above 5 will require a sufficiently powerful graphics card in order to maintain a responsive display.

5.3.4 Particle scaling

Toggles automatic adjustment of particle size with screen size. When enabled, the overall aspect of the display will remain similar even if the view size changes.

5.3.5 Color mode

Provides the following particle-coloring modes:

- Power: the color varies according to the power of the signal in the frequency region
- Dynamics: same as previous except this modes works on signal dynamics
- Power / dynamics: a mix of the above
- Frequency: the color varies according to frequency only, using a rainbow-palette

5.3.6 Power color grading

Determines the range of colors applied to particles, except when using the *Frequency* color mode. Using composite colors, *i.e* not pure red, blue, etc., you will also get whitish tones for high particle values, because of the way RGB layer-blending works.

Particle start color

Color of lowest energy particles.

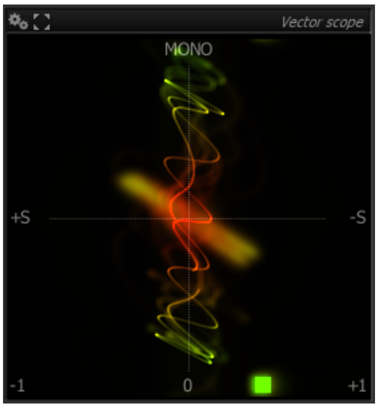
Particle end color

Color of highest energy particles.

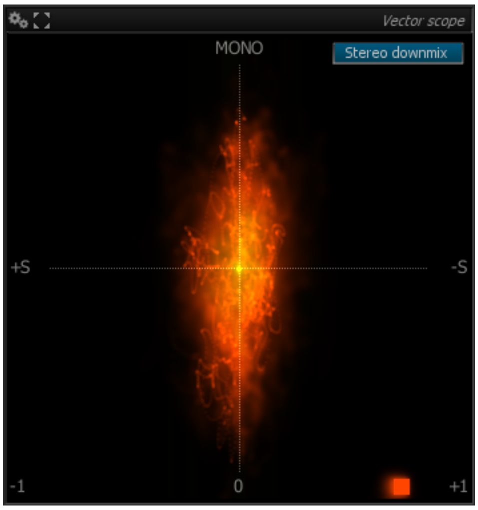
6 Vector scope

6.1 Usage

The vector scope tool is displayed when a stereo input is detected, otherwise the display will switch to Surround scope (see p.48) provided your edition of Pure Analyzer includes this option.



Vector scope display in stereo



Vector scope display in surround (with selection menu)

Modes in Surround :

L-R

Use only Left and Right Channels

Front

Use a stereo down mix with all front channels

Rear

Use a stereo down mix with all Rear channels

Stereo downmix

Use a stereo down mix with all channels

Lt/Rt downmix

Use a Lt/Rt down mix with all channels

LR-Lfe

Use a mono summation of Left and Right + the Lfe (sub) channel

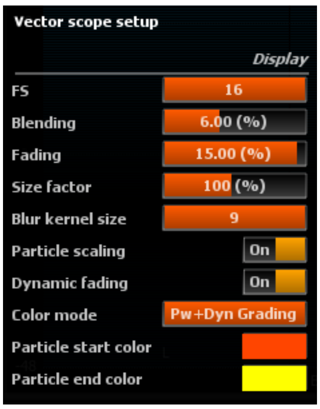
Center-Lfe

Use Center + Lfe (sub) channel

Front-Lfe

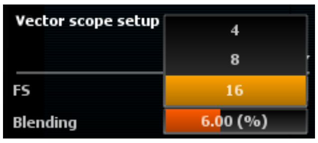
Use a mono summation of the front channels + the Lfe (sub) channel

6.2 Display



Vector scope setup options

6.2.1 Fs



Over-sampling factor in multiples of FS, that is the incoming audio is up-sampled as necessary to reach this multiple times 48kHz. Increasing this value increases the display precision and reactivity, at the expense of a little CPU overhead.

6.2.2 Blending

Controls the amount of particle blending with the current image, from 1 to 100%. A higher value gives more priority to the incoming audio over past frames.

6.2.3 Fading

Controls display persistence, *i.e.* the "fade to black" amount for a frame. Lowering this value retains past particles longer, whereas increasing this make them disappear faster.

6.2.4 Size factor

Controls the size of individual particles with respect to screen size.

6.2.5 Blur kernel size

Controls the radius of the blur effect applied to past particles. Particles are “smeared” more and more as they become older, depending on this setting. Naturally, a bigger value increases the smearing, at the expense of processing power.

Notes

Choosing the value for this setting is really matter of taste, although please keep in mind values that above 5 will

require a sufficiently powerful graphics card in order to maintain a responsive display.

6.2.6 Color mode

This defines how the particle color is determined:

- Static color: use only particle start color (see below)
- Power grading: color is modulated by overall signal RMS power
- Dynamic grading: color is modulated by signal dynamics
- Pw+Dyn grading: mix of the two previous modes

6.2.7 Particle start/end colors

Sets the particle color range to be used.

7 Nebula | Surround scope

7.1 Usage

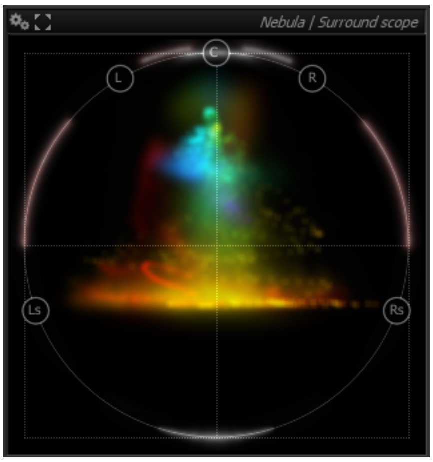
The Nebula | Surround scope displays a representation of how a surround signal's various components are distributed in a surround environment. The inner region displays the location of the signal frequency components in the selected surround configuration, while the outer ring shows the phase-correlation between channels.

Phase correlation between adjacent channels is shown as white section with a length proportional to the correlation. Additionally, L-R phase correlation is displayed on the top portion of the ring, and L-C and C-R inter-channel phase correlations are displayed just above the top of the ring.

Physical locations of the speakers for the selected configuration are marked on the ring itself for reference.

7.1.1 Music

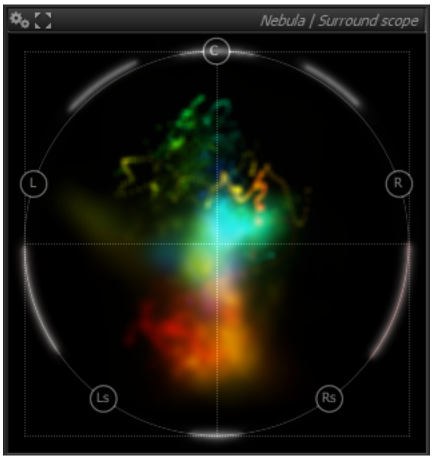
This is the typical surround speaker arrangement for musical reproduction.



Nebula | Surround scope in Music speaker mode

7.1.2 Equidistant

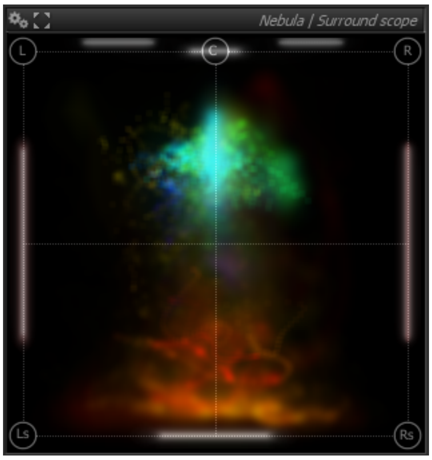
This mode employs equidistant speakers arranged as an equilateral polygon.



Nebula | Surround scope display with equidistant speaker mode selected

7.1.3 Square

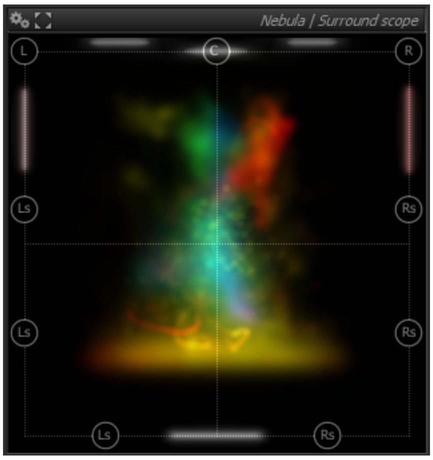
This arrangement employs speakers arranged on a square.



Nebula | Surround scope display in Square speaker mode

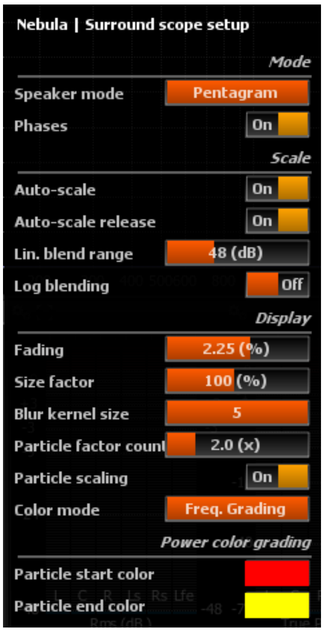
7.1.4 Theater

This is the typical arrangement employed in movie theaters, with redundant rear channels.



Nebula | Surround scope display in Theater speaker mode

7.2 Display



Nebula | Surround scope setup options

7.2.1 Mode

Speaker mode

Selects between various commonly employed surround speaker arrangements.

Phases

Toggles phase-correlation display on and off.

7.2.2 Scale

7.2.2.1 Auto-scale

This parameter controls whether the intensity of the particles are modulated by the overall audio level variations. In essence, when enabled, the color nuances will vary according to the relative amplitude of a frequency, allowing to monitor the relative amplitude spectrum variations. When disabled, the color will reflect the absolute audio level. You can also think of this as a kind of auto-gain setting.

7.2.2.2 Auto-scale release

This controls whether color variations should be smoothed in time or not. When engaged, color variations is slowed down a bit, which makes overall level transitions more obvious.

Notes

You should to enable this setting when you want to visualize quick level variations such as those that frequently occur in movie soundtracks.

7.2.2.3 Linear blend range

Adds a constant blend amount to the particle. This ensures some of the particle is always blended into the image even if its original magnitude is low.

A low value for this setting has the effect of stabilizing the appearance of particles. With large values more of the spectrum dynamics are taken into account, and only peaks mostly come through.

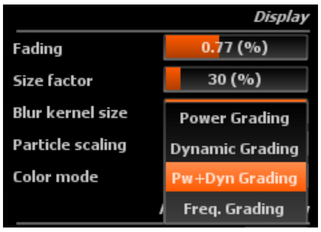
7.2.2.4 Log blending

Toggles between linear and logarithmic blending of the current particle with old particles.

The default is off, ie. linear blending, which tends to favor the display of peaks.

Logarithmic blending on the other hand preserves more of the full dynamic range of the data, and gives some visibility to lower levels also.

7.2.3 Display



7.2.3.1 Fading

Controls display persistence, *i.e.* the "fade to black" amount for a frame. Lowering this value retains past particles longer, whereas increasing this make them disappear faster.

7.2.3.2 Size factor

Controls the size of individual particles with respect to screen size.

7.2.3.3 Blur kernel size

Controls the radius of the blur effect applied to past particles. Particles are “smeared” more and more as they become older, depending on this setting. Naturally, a bigger value increases the smearing, at the expense of processing power.

Notes

Choosing the value for this setting is really matter of taste, although please keep in mind values that above 5 will

require a sufficiently powerful graphics card in order to maintain a responsive display.

7.2.3.4 Particle factor count

Determines the amount of particles to display, relative to the default number used for the current screen size.

7.2.3.5 Particle scaling

Toggles automatic adjustment of particle size with screen size. When enabled, the overall aspect of the display will remain similar even if the view size changes.

7.2.3.6 Color mode

This defines how the particle color is determined:

- Power grading: color is modulated by overall signal RMS power.
- Dynamic grading: color is modulated by signal dynamics.
- Pw+Dyn grading: mix of the two previous modes.

- Freq. grading: color is modulated by frequency.

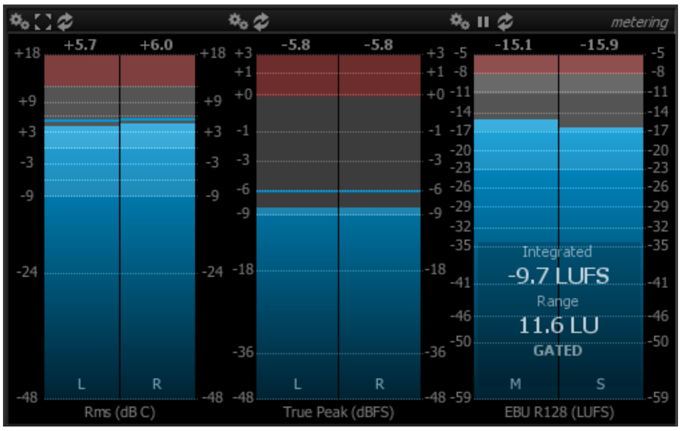
7.2.4 Power color grading

Determines the start and end colors used with ‘Power grading’ color mode selected.

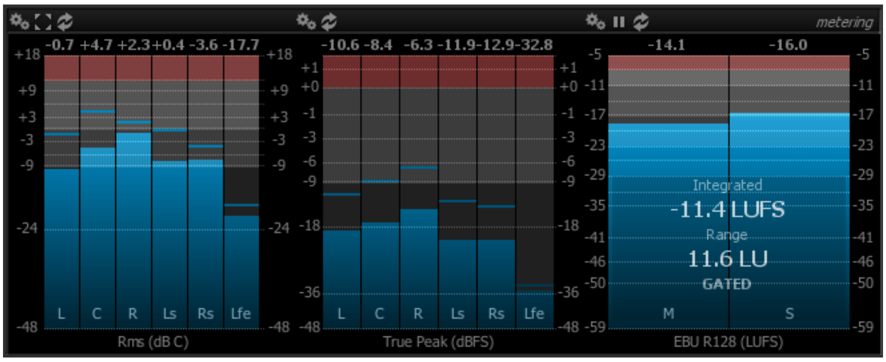
8 Metering

All meters display the current signal meter values as solid vertical bars, and the peaks are indicated with horizontal lines at the corresponding value. Peak hold time can be adjusted in the settings if necessary. The peak value is also displayed in numeric format at the top of the meter, which is emphasized in red in case of clipping or overload.

Several meter displays are available, each scrupulously implementing one of the more common and up-to-date industry norms, as detailed in the following paragraphs.



Meters with stereo input



Meters with 5.1 surround input

8.1 RMS metering

8.1.1 Introduction

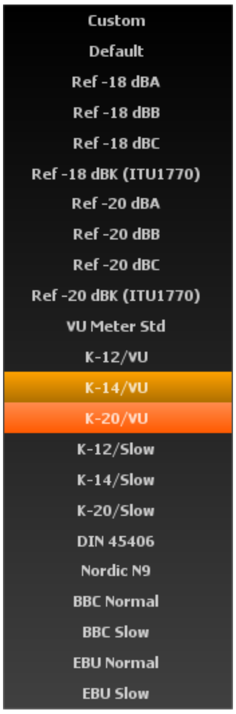
RMS stands for Root Mean Square, is a measure of the average magnitude of a varying signal, or equivalently, the average power over the signal over a time period, called the integration time.

Notes

The live layouts display dB SPL (Sound Pressure Level) values, which is the standard measure of acoustic pressure. This requires that your input chain first be calibrated in order to get accurate and meaningful readings, as factors such as your particular microphone's sensitivity and preamplifier gain are not known in advance. For this, you will need to get your hands on a calibrator, which is a box fitted with a transducer that outputs a known acoustic level and features a socket designed to hold the microphone.

8.1.2 Preset

A number of presets covering widely and not so widely-used metering standards are provided.



Available RMS metering presets

Custom

User defined values.

Default

All-round settings with:

- From -48 to +18 dB range, referenced at -18dB.
- 160ms integration time, 16dB/s release, 1dB peak release and 60 frames peak hold.

Ref -18dB A/B/C/K

Default settings with pre-equalization following either normalized ANSI A/B/C or ITU-R BS.1170-2 weighting curves, referenced to -18dB.

Ref -20dB A/B/C/K

Default settings with pre-equalization following either normalized ANSI A/B/C or ITU-R BS.1170-2 weighting curves, referenced to -20dB.

VU meter Standard

Standard reference vu settings, with 300ms integration, 66/7dB/s release and peak release times, referenced at 0VU/-4dBu/-18dBFS. The scale is non-linear and covers -20 to +3VU, complying with IEC 60268-17.

K-System / VU

Linear scale, conforming to Bob Katz's recommendations, referenced at either -12, -14 or -20dB. 300ms integration, 66.7dB/s release and 12dB/s peak release times, 180 frames peak hold.

K-System / Slow

Identical to K-System/VU, except that integration times are doubled. This reflects Bob Katz's view that Vu-meter timings are appropriate for speech, but that longer timings are better suited to music.

DIN 45406

This preset conforms to the standard used many European broadcasters such as French (PAD) and German (IRT) television. Integration time is 10ms for a 90% signal increase; fall-back time is 1.7s per 20dB; with a linear scale covering a range from -50 to +5dB, referenced at -9dBFS. The corresponding standards are DIN 45406, IEC 60268-1, and ARD Pfl.H.3/6.

Nordic N9

5ms integration time for an 80% increase, fall-back time 1.7s per 20dB, linear scale covering the range from -40 to +9dB, referenced at -18dBFS, according to IEC 60268-10/1 + N9 supp.

BBC Normal

10ms integration time for an 80% increase, fall-back time 2.8s per 24dB, custom scale with graduations spaced apart by 4dB, and 4 stands for the -18dBFS reference, according to IEC 60268-10/2a.

BBC Slow

Same as above except for ballistics, where the integration time is changed to 69.2ms for an 80% increase, and 3.8s per 24dB fall-back.

EBU Normal

10ms integration time for an 80% increase, fall-back time 2.8s per 24dB, linear scale covering the range from -12 to +12dB, referenced at -18dBFS, according to IEC 60268-10/2b.

EBU Slow

Same as above except for ballistics, where the integration time is changed to 69.2ms for an 80% increase, and 3.8s per 24dB fall-back.

8.1.3 Reference

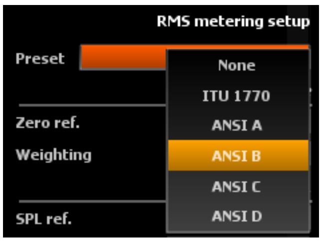
RMS metering setup options

Zero reference

Adjusts the reference point. Default is -18dB (DVD standard). Do not change this unless you specifically want to divert from the standard, as this will otherwise compromise meter readings.

Standard values are -18dB for DVD authoring and -20dB for film.

Weighting



Applies an optional weighting filter conforming to various standard curves:

- None (default).

- ITU 1770: K-weighting filter, comprising of a shelving and a high-pass (RLB-weighting) filter in series, as specified in ITU-R BS.1170-2 and employed by EBU R128 (PLOUD).
- ANSI A, which is roughly the inverse of the Fletcher-Munson curve.
- ANSI B.
- ANSI C.
- ANSI D.

8.1.4 SPL

SPL reference

This is the reference level of the calibrator's output, indicated on the device itself or in the corresponding datasheet. A typical value is -94dB.

SPL trim

This is the offset applied to RMS dB values in order to obtain dB SPL readings. It is determined automatically by the calibration procedure.

Calibrate

Press this button after having insert the microphone into the calibrator socket and activated it in order to determine the SPL trim value.

8.1.5 Range

8.1.5.1 Min / max

Defines the minimum and maximum values to be displayed on the meter bars. This does not affect the text readings above the bars.

8.1.6 Time

Integration

Defines the meter integration time constant, in milliseconds. This corresponds to the length of the time window over which an RMS level value is computed. Decrease this to respond to signal level variations more quickly, at the expense of meter precision and stability. Default is 160ms.

Release

Release time of the meter, in decibels per second. This controls the falloff rate of the meter. Decrease this to respond to signal level variations more quickly, at the expense of readability. Default is 16 dB/s.

Peak release

Release time of the peak indicator, in decibels per second. This controls the falloff rate of the peak hold indicators. Increase this to retain peaks for a longer time. Default is 1dB/second.

Peak hold

Sets the number of display frames to wait until the peaks actually start to fall-back to zero. Default is 60 frames.

8.1.7 Scale & split

Scale

Meter labels are defined here as a comma separated list of dB values to be shown on the side of the meters. This also defines where to the corresponding horizontal markings. Default is -72;-40;-18;-9;-6;-3;-1;0;1;3.

Colors

This lets you customize the values at which color transitions occur. You can enter as many values as you wish, as a comma separated list, but make sure the values are in increasing order. Default is -9;0.

The last value always defines the clip level, which will be indicated in red.

8.1.8 Other

8.1.8.1 Bar-graph texturing

Controls wether meters are drawn with texture or in a plain solid color. Default is on.

8.2 True Peak metering

All digital audio wave signal is ultimately converted back to analog at some point, and while it is often desirable to maximize the overall volume of a signal or a complete mix, care must be taken in order not to go above the digital scale zero decibel ceiling, or nasty distortion and clipping will occur. This common and widely used rule is however not entirely sufficient, as the digital and analog processing involved in a D/A converter does not guarantee that a 0dBfs peak signal will exactly translate to a 0dB peak in the analog domain.

Without getting into too much detail, this phenomenon can be attributed to the over-sampling and reconstruction filters present in D/A converters, whose role are to rebuild a continuous time signal from a set of discrete digital values sampled at regularly spaced time intervals. This interpolation process can therefore generate values which lie above 0dB, which is known as overshoot.

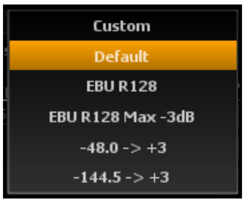
Relying solely on the peak value of samples can lead to the following problems:

- Inconsistent readings between successive playbacks of the same material.
- Unexpected overloads of the D/A output converter.
- Under-readings and beating of pure tones.

TruePeak metering aims to overcome these limitations by mimicking parts of the D/A conversion process, effectively up-sampling the measured signal, in order to display the true value of peaks that occur in the analog domain.

TruePeak metering setup options

8.2.1 Preset



Custom

User defined values.

Default

This preset uses the following all-round settings:

- Range: -72 ... +3 dB referenced at 0dB.
- Scale: 1.8x power factor, 0.06x reference display offset.
- Ballistics: 16dB/s release time, 1dB/s peak release, 60 frames peak hold.
- Scale / split: -72, -40, -18, -9, -6, -1, 0, +1, +3 dB.

EBU R128 Referenced at -1dB:

EBU R128 Referenced at -3dB:

-48.0 -> +3

Limited -48 ... +3dB range with adapted scale/split values.

-144.5 -> +3

Wide -144.5 ... +3dB range with adapted scale/split values, to monitor the full 24-bit dynamic range and possible clipping.

8.2.2 Range

8.2.2.1 Min / max

Defines the minimum and maximum values to be displayed on the meter bars. This does not affect the text readings above the bars.

8.2.2.2 Ref

Controls the position of the reference value on the display. This does not affect the meter values per se, it is a cosmetic setting only.

8.2.3 Scale

Power factor

Controls the scaling of the display with respect to meter values. This allows to stretch or compress the display around Reference.

Ref pixel offset factor

Adjust the offset of the reference value (Reference) with respect to meter height.

8.2.4 Time

Release

Release speed of the meter in decibels per second.

Peak release

Release speed of the peaks in decibel per second.

Peak hold

Number of frames to hold the peaks for, before the actual release phase begins. 60 frames corresponds to 1 second on a fast system, capable of a 60Hz refresh rate.

Infinite hold

When enabled, peaks are held until the next reset, which is useful for checking a whole track never clips.

Reset

Clicking the button resets the meter to its initial state (values and peaks at minimum).

8.2.5 Scale & split

Scale

Meter labels are defined here as a comma separated list of dB values to be shown on the side of the meters. This also defines where to the corresponding horizontal markings. Default is -72;-40;-18;-9;-6;-3;-1;0;1;3.

Colors

This lets you customize the values at which color transitions occur. You can enter as many values as you wish, as a comma separated list, but make sure the values are in increasing order. Default is -9;0.

The last value always defines the clip level, which will be indicated in red.

8.2.6 Other

Controls wether meters are drawn with texture or in a plain solid color. Default is on.

8.3 Loudness: ITU-R BS. 1770 and EBU-R128 (PLOUD)

ITU-R BS.1170-2 and EBU R128 recommendations introduce a new paradigm for audio metering, which define a way to measure perceived loudness of audio material in a normalized and reproducible manner.

Please refer to the official documents freely available online at http://tech.ebu.ch/groups/ploud or consult a reference book such as “Audio Metering. Measurements, standards and practice” by Eddy Brixen (Focal Press, ISBN 9780240814674) for detailed information on this subject.

8.3.1 Principles

8.3.1.1 Units

ITU-R BS.1170-2 notably defines LU (Loudness Unit) and LUFS (Loudness Unit, referenced to Full Scale) units, which are used by EBU R128, and Maximum True Peak Level.

- LU is used for measurements *relative* to a reference level and measuring range.
- LUFS is used for *absolute* measurements.

The meter display is switchable between LUFS (absolute, default) and LU (relative). The target loudness level to aim for is -23 LUFS = 0 LU.

8.3.1.2 Loudness and EBU mode

EBU mode specifies three time scales corresponding to three different, complementary loudness levels

- M: Momentary, 400ms integration time
- S: Short-term, 3s integration time
- I: Integrated from start of measurement or last reset, gated

Please note that loudness is a measure of global loudness, so individual channel metering is not relevant in this context. No additional slowdown of the attack or release of the meter is employed, as indicated by the norm.

The integrated loudness can be understood as the overload loudness of the audio over time, excluding very soft passages through the use of absolute and relative gating.

8.3.1.3 Loudness Range (LRA)

Loudness range measures the average long-term variations of the loudness; it is expressed in LU.

8.3.1.4 Scales

EBU R128 specifies two normalized scales:

- EBU +9, ranging from -18.0 LU to +9.0 LU (-41.0 LUFS to -14.0 LUFS)
- EBU +18, ranging from -36.0 LU to +18.0 LU (-59.0 LUFS to -5.0 LUFS) (Default)

8.3.2 Dolby Dialogue Intelligence™

Introduction

While EBU R128 aims to measure global perceived loudness, irrespectively of the audio material, Dolby Dialogue Intelligence is a patented technology designed to specifically measure the perceived loudness of dialogue elements in the audio. It is therefore targeted towards broadcast applications.

General principle

Dialogue Intelligence replaces EBU R128's level-based gate with a speech-content ratio based gate. The algorithm computes several low-level features for the incoming signal in speech channels. These are then combined into an overall speech percentage figure. When speech content is detected, Integrated Loudness is computed from the speech channels which have a speech content ratio above a certain threshold (see Dolby Dialog Intelligence (see p.67)).

When other material is detected, *i.e.* not speech, standard EBU R128 Integrated Loudness computation is employed.

Display

The current speech content is displayed as text below the current gate status.

Additionally, color coding indicates the speech content ratio.

- Speech : speech content present
  - **Green** : high speech content
  - **Orange**: medium speech content
  - **Red**: low speech content
- **Other**: other material present

Delay and compensation

The sophistication of the algorithms employed in Dialogue Intelligence incur an overall latency of 2048ms (approx. 2s).

When Dialogue Intelligence is enabled, the display of other Loudness values is compensated to make sure meter readings are consistent. Other real-time meter (RMS, TruePeak) displays are not compensated, as we feel in this case maintaining the best reactivity to the incoming signal is more important.

All meter statistics are time-aligned.

Surround

Channels taken into account by the algorithm are determined based on the current channel configuration.

For mono/stereo signals, all channels are taken into account. For surround configurations, only Left/Right and Center channels are taken into account, if present.

Notes

Dialogue Intelligence computation only affects I (Integrated) Loudness values.

Toggling Dialogue Intelligence on and off forces a reset of the meter values.

8.3.2.1 Copyright & patent information

Created under license from Dolby Laboratories Licensing Corporation. Use of this Software does not convey a license nor imply a right under any patent, or any other industrial or intellectual property right of Dolby Laboratories.

Dolby and the double-D symbol are registered trademarks of Dolby Laboratories. Dialogue Intelligence is a trademark of Dolby Laboratories.

PATENT LIST - DIALOGUE INTELLIGENCE

PATENTS

Country	Patent Number
AUSTRALIA	2003263845
CHINA	ZL03819918.1
FRANCE	1 532 621
GERMANY	1 532 621
HONG KONG	1073917
ISRAEL	165938
JAPAN	4585855
MEXICO	252,228
MALAYSIA	MY-133623-A
SINGAPORE	109865
TAIWAN	I306238
UNITED KINGDOM	1 532 621
UNITED STATES	7,454,331

PATENT APPLICATIONS

Country	Application Number
CANADA	2,491,570
INDIA	1936/KOLNP/2004
SOUTH KOREA	2005-7003479
UNITED STATES	12/948,730

8.3.3 Controls and display

Display

The meter display has the following arrangement

- left bar: Momentary Loudness value
- right bar: Short-term Loudness
- text overlay: Integrated Loudness and Loudness Range (LU) values, Gated indicator lights red when gate is active

Pause

Clicking the button pauses measurement; clicking again resumes it. This allows you to make adjustments without affecting Integrated Loudness, instead of having to start all over again.

Reset

Clicking the button resets the meter to its initial state.

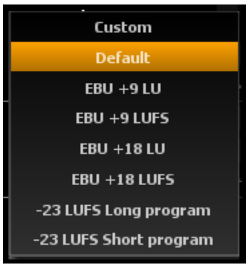
Notes

Don't forget to reset the Loudness meter if you're starting playback of a new track, as Integrated Loudness, by design, measures the overall Loudness since the last reset. Otherwise you'd be measuring the overall Loudness of the combined tracks, which is probably not what you want.

8.3.4 Setup

EBU R128 Loudness metering setup

8.3.4.1 Presets



Custom

Use user-defined custom range according to min./max. values below.

Default

Sets the meter to the recommended scale (EBU +18 LUFS).

EBU +9 LU

Sets the meter to use EBU +9 scale in LU units.

EBU +9 LUFS

Sets the meter to use EBU +9 scale in LUFS units.

EBU +18 LU

Sets the meter to use EBU +18 scale in LU units.

EBU +18 LUFS

Sets the meter to use EBU +18 scale in LUFS units.

-23 LUFS Long program

CST specification

Sets the meter to use EBU +18 scale in LUFS units with reference @ -23 LUFS and color split @ -/+ 7LU from the reference.

-23 LUFS Short program

CST specification

Sets the meter to use EBU +18 scale in LUFS units with reference @ -23 LUFS and Max defined 3LU up to the reference.

8.3.4.2 Dolby Dialogue Intelligence

Dolby Dialogue Intelligence™

Toggle usage of Dolby Dialogue Intelligence speech gate.

Speech threshold

Defines the speech content threshold in %. Speech channels with a speech content ratio below this value do not participate in the Loudness computation.

8.3.4.3 Range

Min.

Minimum Loudness to display on the bar-graphs. User adjustable.

Max.

Maximum Loudness to display on the bar-graphs. User adjustable.

8.3.4.4 Scale / split

Scale

Meter labels are defined here as a comma separated list of dB values to be shown on the side of the meters. This also defines where to the corresponding horizontal markings. Default is -72;-40;-18;-9;-6;-3;-1;0;1;3.

Colors

This lets you customize the values at which color transitions occur. You can enter as many values as you wish, as a comma separated list, but make sure the values are in increasing order. Default is -9;0.

The last value always defines the clip level, which will be indicated in red.

8.3.4.5 Other

Controls wether meters are drawn with texture or in a plain solid color. Default is on.

8.4 Leq metering

8.4.1 Introduction

Leq encompasses a set of sound level meter specifications, which are described in detail in the BS EN 61672-1 European Standard.

Pure Analyzer implements the following Leq measurements: time-weighted sound level, time-average sound level and sound exposure level.

Frequency weighting is employed for all measurements, A being the standard and default, although other weightings can be specified if necessary.

The Leq module always measures the audio routed through the Mic channel

8.4.1.1 Time-weighted sound level

L_A is the root-mean-square sound level obtained after exponential time weighting.

Exponential averaging has the effect of progressively 'forgetting' past sample values.

The norm specifies two time-weighting constants

- Fast : 125ms
- Slow: 1s

The corresponding letter symbol is L_{AF} for a an A-frequency weighted and F time-weighted sound level, for example.

8.4.1.2 Time-average sound level

Time-average sound level is basically a RMS meter with frequency-weighting applied.

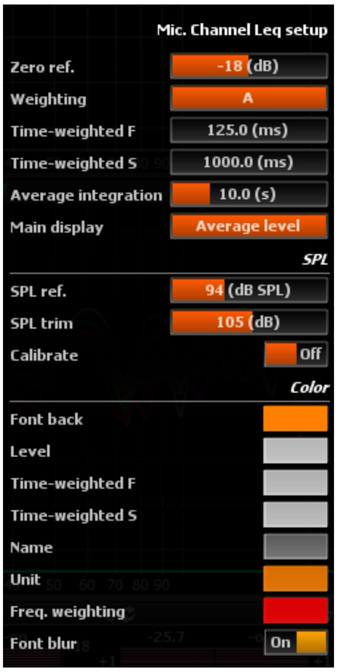
8.4.1.3 Sound exposure level

This measures the sound exposure equivalent to a 'dose' received for a second. It is useful for determining the amount of sound pressure to which listeners have been exposed for a certain duration.

Notes

This value naturally increases with time. For a constant source level, this value increases in a logarithmic fashion.

8.4.2 Mic. channel Leq setup



Mic. channel Leq setup

Zero ref.

Adjusts the reference point. See RMS (see p.57) for more information.

Weighting

Frequency weighting employed. Can be switched between ANSI standard (A, B, C, D) and none. The default is A.

Time-weighted F

Indicates the time-constant for the *Fast* time-weighted sound level.

Time-weighted S

Indicates the time-constant for the *Slow* time-weighted sound level.

Average integration

Sets the integration time for the time-average sound level, between 1s and 14400s (4 hours). Default is 10s.

Main display

Switches the main measurement display from time-average sound level (the default) to sound exposure level.

8.4.3 SPL

SPL reference

This is the reference level of the calibrator's output, indicated on the device itself or in the corresponding datasheet. A typical value is -94dB.

SPL trim

This is the offset applied to RMS dB values in order to obtain dB SPL readings. It is determined automatically by the calibration procedure.

Calibrate

Press this button after having insert the microphone into the calibrator socket and activated it in order to determine the SPL trim value.

8.4.4 Color

The following settings control the visual aspect of the Leq display.

Font back

Common font background color

Level

Main level font color

Time-weighted F

Fast time-weighted level font color

Time-weighted S

Slow time-weighted level font color

Name

N/A

Unit

Unit display font color

Freq. weighting

Frequency weighting type display font color

Font blur

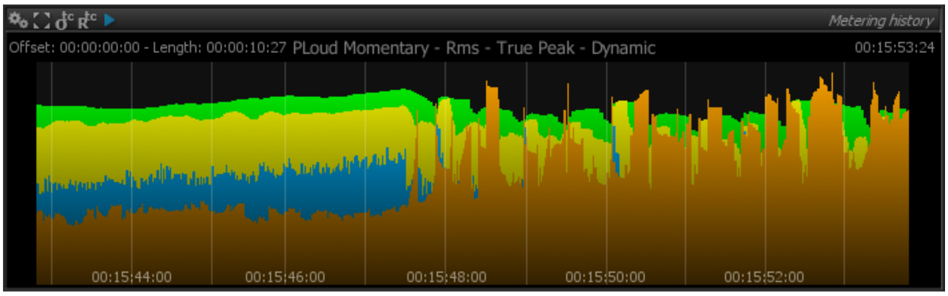
Toggles font blurring on (default) and off.

9 Metering history

9.1 Usage

The metering history panel stores and displays the evolution of meters over time, with a red vertical bar indicating current time. Start and end time-points of the period over which the history are displayed left and right in time-code format.

Selecting which meters are to be included in the display is done by clicking the corresponding buttons in the setup.



Metering history display

9

TimeCode offset

Clicking the button defines the current time as the TimeCode offset.

TimeCode offset reset

Clicking the button resets the TimeCode offset to zero. Absolute and relative TimeCode will then be the same.

Play

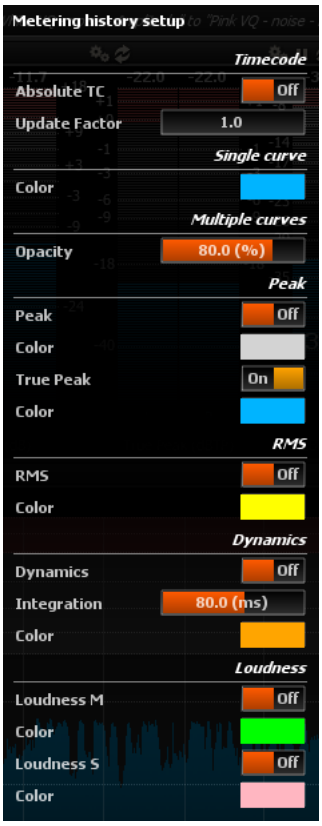
Clicking the toggles history recording on and off. Metering values are discarded when off.

Notes

The metering history relies on the same settings as those defined in the various meters. However, when multiple meter values are displayed simultaneously, the display range of the history is adapted so it encompasses the display ranges of the meters.

Keep in mind different meters can be set to different zero reference points when comparing meter history curves.

9.2 Setup



Metering history setup options

9.2.1 TimeCode

Absolute TimeCode

Switches between absolute and relative TimeCode formats.

Update Factor

Divide the History refresh interval; allowing to increase the history time period.

9.2.2 Single curve

Color

Sets the color to use when only a single curve is selected for display.

9.2.3 Peak

These settings allow to specify whether Peak and/or TruePeak curves should be displayed, as well the color to use when drawing them.

9.2.4 RMS

Toggle RMS curve display on and off, and specify the color to use for drawing.

9.2.5 Dynamics

The dynamics the current dynamic range of the signal, that is the ratio of the peaks with respect to the average, *i.e.* the crest factor of the signal.

Dynamics

Toggles dynamics curve display on and off.

Integration

Set the integration time, in milliseconds.

Color

Specify the color to use for drawing the curve.

Remarks

Percussive content such as drums or rhythm guitar exhibit high dynamics, as opposed to sustained sounds such as strings and synthesizer pads.

9.2.6 Loudness

These settings allow to specify whether Short-term and/or Momentary EBU R128 Loudness curves should be displayed, as well the color to use when drawing them.

10 Metering statistics

The metering statistics view shows a synthetic view of the current and past meter values in numeric form. It also serves to process multiple existing audio files in one pass, display and export the results to disk.

Metering statistics display

10

10.1 Audio level statistics

10.1.1 Overview

The display shows the average and range for the various level meter values, since the start of the application or the last time the meter was reset, in a spread-sheet type view.

Peak, TruePeak and RMS

Mean as well as overall minimum and maximum values are shown. For min. and max. values, the corresponding TimeCode position is also indicated.

Loudness

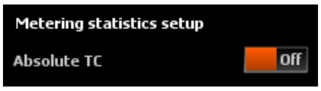
As EBU R128 Loudness already incorporates statistical computations, only the current values are shown.

10.1.2 File export

Exports a report containing a summary of the metering statistics data to a text file.

Clicking the button brings up a standard file dialog where you can specify the desired file name for the dialog.

10.1.3 Setup



Absolute TimeCode

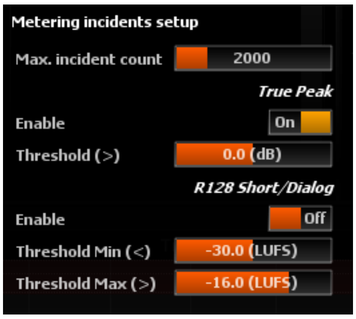
Toggles between relative and absolute TimeCode display. See TimeCode (see p.72) for more information.

10.2 Incident reporting

10.2.1 Overview

All TruePeak and R128 Short term values that cross the thresholds are recorded and displayed as a list. Each row in the list shows a record of the offending peak value in dB alongside with the time-code at which the event occurred. You can navigate the list and locate the time positions of the incident, then playback again the corresponding source material in order to identify and correct the problem.

10.2.2 Setup



Incidents setup options

Max. incident count

To avoid overloading the display, and eventually, the computer's memory, there is a limit placed on the number of registered incidents, which is 2000 by default. If you go above this, it might be a good idea to back off the master fader a bit anyway to let that music breathe !

However, you can override this behavior by setting this value to -1, which will remove the limit altogether.

TruePeak Incident Enable + threshold

Defines the threshold above which an incident will be registered. Default is 0dBTP, which corresponding to full digital scale. A conservative value would be -0.1dBTP, to be on the safe side.

Keep in mind TruePeak is designed to measure inter-sample peaks, and that 0dBTP is actually a few tenths of decibels softer than digital peak.

EBU R128 Short term / Dialog Incident Enable + thresholds

Defines the threshold under/above which an incident will be registered.

10.3 Off-line processing media queue

10.3.1 Usage

Multiple audio files can be added to the list for unattended queue processing.

Principle

The media queue is intended for processing a soundtrack possibly split across several reel and channels. Reels are processed in the order in which they are added and in which they appear in the list.

Usage

Audio files are added by clicking the icon, which brings up a standard file selection dialog, where you can select as many files as you want, provided they all have the same channel count and in a supported format, with a recognized extension (.wav). When you are ready, click the icon to start processing the list, which will be computed much faster than real-time, especially if you have a fast computer.

The results are displayed when ready in the main view, and you can export these to a file just as you would with metering statistics computed on incoming audio.

Reel grouping

If reels are split across several multi-channel files, you can add all the files at once directly in the file selection dialog. Reel order corresponds to the order in which the files were added.

Channel grouping

If your source material consists of mono-tracks, you must add reels one-by-one, adding all files for the various channels of the current reel. Please ensure different reels have the same channel count or the software will report an error. Channel configuration and names are inferred from the file names using a fuzzy-logic algorithm that looks for the presence of typical marker characters such as C / Center for the center channel, R / Right for the right channel etc. (case insensitive).

If the automatic channel grouping does not succeed, an error message will be displayed. Please rename the offending file(s) according to one of the expected schemes above to correct the problem.

Notes

This function is not intended to process unrelated soundtracks in batch mode. Please repeat the operation as necessary if you wish to obtain separate results for individual tracks.

11 System analysis

11.1 Introduction

At first glance, an audio signal chain is very much like a series of black boxes. As an audio engineer, you can trust your ears and the manufacturer's data-sheets to assess the effects this chain has on the incoming audio. In a variety of cases, however, this is either simply impractical, not possible or not precise enough. Such situations include live sound setups, recording setups, etc., where unknown factors such as the venue's or studio's acoustic response are a crucial part of the chain.

It is therefore necessary to resort to scientific measurement procedures and tools to obtain precise, trustworthy and reproducible results. The main tools at your disposal for this purpose are transfer curve and impulse response measurement, which are especially designed for this task.

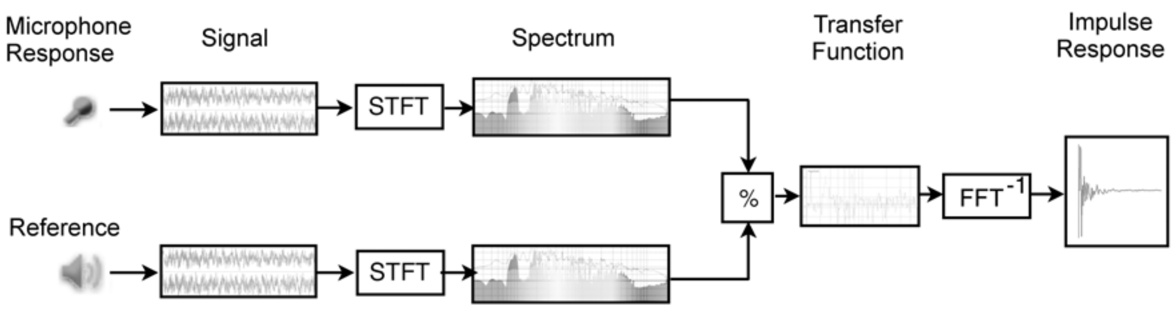
As with any measurement instrument, it is important to have a good grasp of its mode of operation as well as any possible limitations in order to use it most efficiently. Some knowledge of acoustic principles and notions of signal processing are naturally required as well. While this manual tries to cover most typical use cases and point out common do's and dont's, it obviously cannot replace neither a good textbook nor practical experience.

11.2 Initial setup

Throughout this manual we will refer to the measured signal processing chain as the *system* (sometimes called device under test in electronics literature). This system input is fed with a *source* signal, which produces a *response* signal at its output(s). Both source and response are recorded and monitored by the analyzer, from which several measurement curves are produced.

The first step is therefore to setup the measurement chain. In case an outboard or plugin device's characteristics are to be measured, this is just a matter of routing the inputs and outputs in your DAW.

If you're measuring the acoustic response of a physical space, you'll need to place at least one microphone at the preferred listening position to record the response. The source can either be picked up directly at the DAW output or recorded with a second microphone placed in front of the loudspeaker(s), depending on whether you want to include the loudspeaker's influence or not in the measurement.



System analysis overall principle

11.3 Practical considerations for capturing measurement signals

Use a measurement microphone

The goal here is to take the measurement chain out of the equation, so only specially designed microphones that exhibit a flat curve, minimal coloration, lowest noise and distortion should be used.

Choose a neutral preamplifier and calibrate it accurately

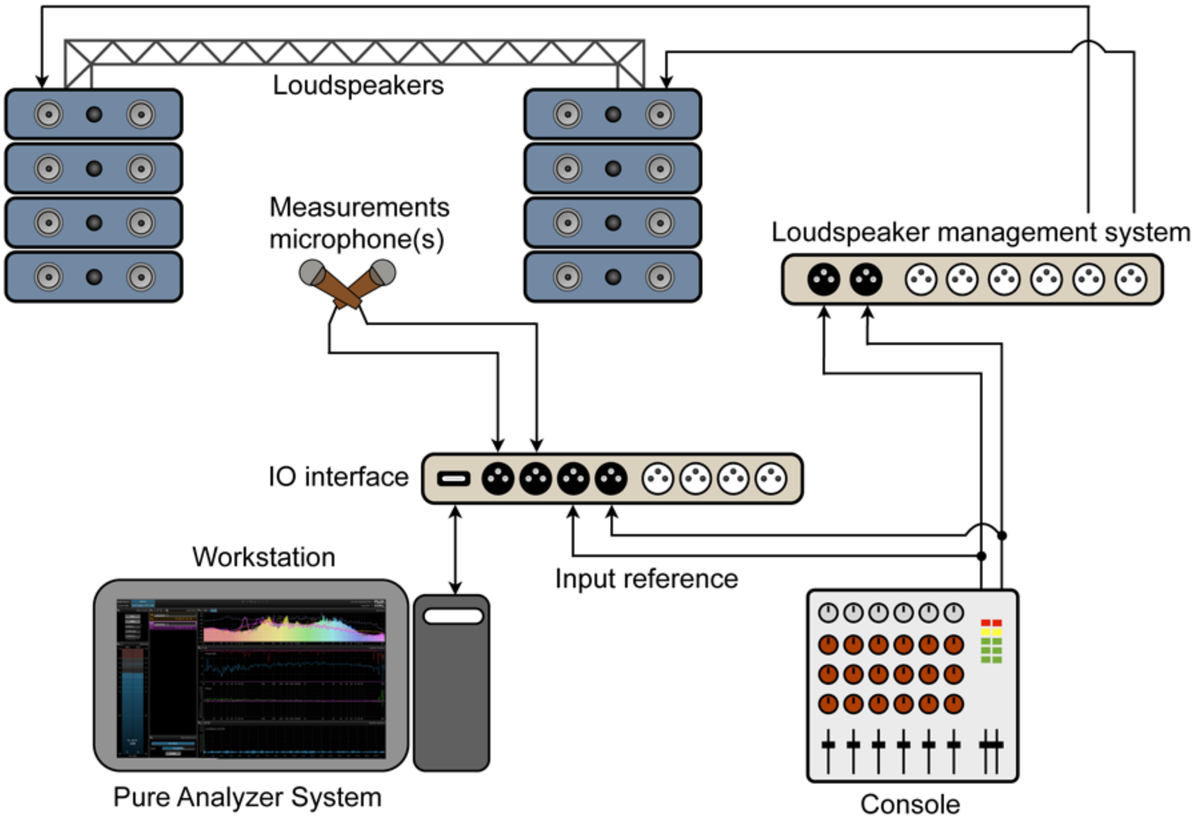
For the same reasons, select the most neutral preamplifier and A/D D/A converters you have at your disposal. It is especially important to be able to set accurate and reproducible gain, linear and flat response. Take special care that the signal is not so hot as to clip or distort the preamplifier input stages, as this would distort the measurements accordingly and induce you into error.

Maximize signal-to-noise ratio

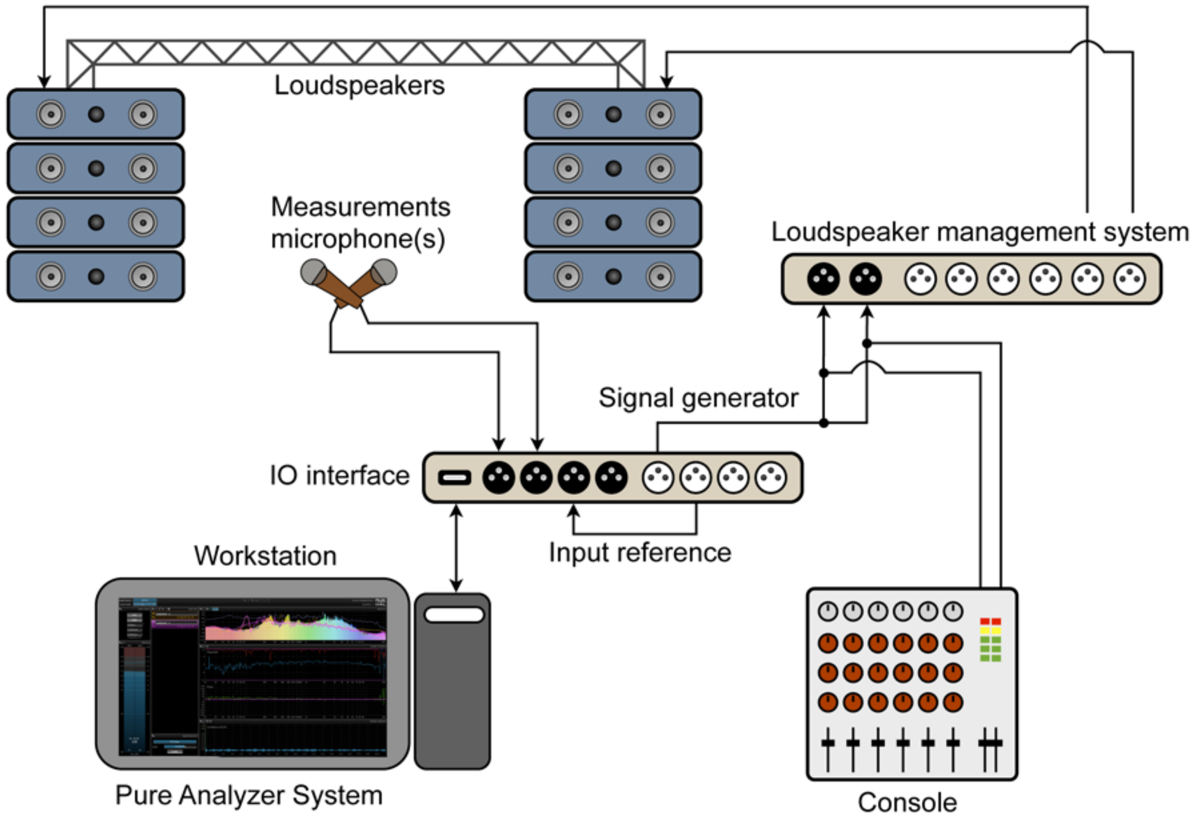
When measuring an acoustic system, raise up the volume as high as practical for maximal signal-to-noise ratio, and try to minimize any spurious acoustic noises such as footsteps and conversation. As always, the goal is to set the test signal as high as possible above the noise floor while ensuring all devices still operate in their linear region. Finally, make sure the microphone is firmly held in position and acoustically decoupled from the floor.

In a live concert context, especially with the audience present, using a noise signal is not practical. In this case you can still perform measurements, using a music signal, but the measurements will be less accurate as the signal isn't known in advance and does not necessarily contain all frequencies like noise does.

11.4 Measurement setup



Typical configuration for a live venue measurement setup using external signal generator



Typical configuration for a live venue measurement setup using Pure Analyzer's internal signal generator and loopback

11.5 Test signals

Pure Analyzer is designed to cover the broadest range of practical use cases, and does not impose a limitation on the measurement signal used.

Traditionally, transfer curve and impulse response measurements are performed by feeding a specially designed test signal into the system, the most commonly employed being pink and white noise and swept sines. While these type of signals are those that give the best and more accurate results, with each having its own strength and weakness, they do prohibit the measurement of a system in the context of a live system with the audience present.

Performing measurements using a live music signal allows the engineer to fine tune the system settings to compensate for changing conditions such as the effect of the crowd on acoustic reflections and damping, varying temperature and humidity, etc. Although less pleasing to the ear, we do however recommend using a noise test signal whenever possible, at least as a starting point.

You are free to use any kind of test signal generator, outboard or plugin, provided you trust it being reliable and easy to use. A selection of plugins suitable for this task is shown in the chart below.

Remarks

While Pure Analyzer does not impose any limitation on the test signal used, we recommend using the integrated Signal generator (see p.87), which has been especially designed for this task. We conducted thorough tests on a wide panel of signal generators available as plugins or integrated into DAW software and found that many do not meet the requirements for performing accurate and reliable measurements.

12 Live IO

12.1 Introduction

The delay finder's role is to determine the total delay of the signal path, from source to response. Note that this excludes any delays induced by your soundcard and DAW, as these should be compensated for and equivalent to zero as explained before. Here we are only concerned with the time taken by sound-pressure waves to travel the distance from loudspeakers to the measurement microphone placed at the listener position.

This figure must be determined with sample accuracy in order to establish proper transfer function and impulse response measurements. In a sound installation context, computing precise time-delay is crucial to align multiple speakers and transducers properly, as to minimize comb-filtering artifacts.

12

12.2 Basic operation

Compute the delay

Press the button to find the delay using the most recent incoming audio. The resulting figure is displayed almost instantly as a:

- Delay in samples (smp).
- Distance in meters (m) or Imperial feet (ft.) depending on wether Metric system (see p.17) is enabled.
- Delay in milliseconds (ms).

Compensate the delay

Pressing the button activates a delay line in the the source signal path to compensate for the currently displayed delay value, effectively time-aligning *source* and *response* signals. Pressing again deactivates the delay line, which allows for quick comparison between uncompensated and compensated signal paths.

Fine-tune manually

If necessary, you can manually adjust the delay figure using either of these methods:

- Direct keyboard numeric value entry as time or distance figure.

- Increment / decrement by clicking the / icons.
- Increment / decrement using the +/- numeric keys.

Perform a new measurement

Press the button again to perform a new measurement. This will overwrite any previous value.

12.3 Notes

Max. delay time and room/venue size

The maximum measurable delay time is adjustable in the settings. Attempting to measure a delay greater than this will inevitably lead to corrupt measurements. The default setting is 1s, which should cover the vast majority of real-world situations, since it covers a distance of 330 meters.

Ensure stable conditions while performing a measurement

You should ensure both source and response signals have reached have reached stability before attempting measurement. In particular, do not stop or start the audio, change the volume or any other parameter just before or during measurement. This would invalidate the measurement and you would have to start again.

Limitations

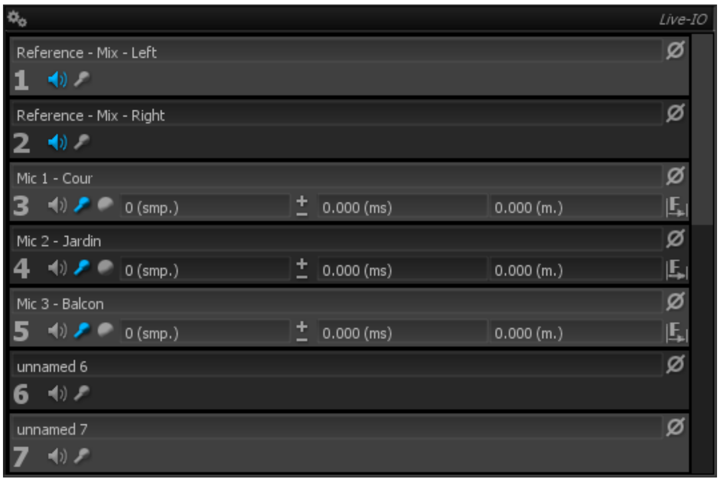
Please note there are many unknowns in play when determining the optimum delay figure. While we did our best to make this tool as robust and accurate as possible, as with all automatic procedures there is always a possibility that it will fail. In this case you should repeat the process or resort to manual adjustment until you get satisfactory results.

Multiple paths

The major assumption behind delay compensation is that there is a main direct path from source to listener. In a very reverberant or complex-shaped acoustic space, this obviously does not apply anymore. This is where acoustics expertise and trial-and-error comes into play, in order to attain the best compromise.

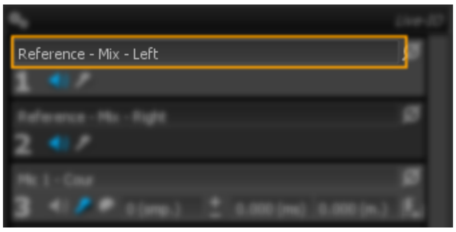
12

12.4 User interface and controls



Live IO controls

12.4.1 Name



Allow to define a custom name for each channel. This is a global name; saved and restored with the preset but not directly related to the Hardware I/O Interface. As this, it will be consistent even if you switch the Hardware I/O Interface or switch to connect to a SampleGrabber.

12.4.2 Ref

The button toggles wether the corresponding channel should be used as a reference signal.

Multiple channel can be used as reference, in which case a mono-sum of these channels is used as the internal reference signal.

12.4.3 Mic

The button toggles wether the corresponding channel should be used as a microphone input signal, which is used to capture the response of the system.

Multiple channel can be used as microphone input, in which case a mono-sum of these channels is used as the internal microphone signal.

12.4.4 Phase invert

The toggles phase inversion of the selected channel on and off. This can be used to compensate another known phase inversion somewhere else in the analog signal chain.

12.4.5 On/Off

The button toggles delay compensation on and off. When the correct delay has been determined, engage this button to insert a delay line in the reference channel, to align reference and measured signals, and get correct transfer function and impulse response.

12.4.6 Delay value

The delay value is displayed simultaneously as:

- A number of samples (at the current sampling rate).
- A time delay, in milliseconds.
- A distance, in meters or feet.

You can manually adjust any of these values, using either keyboard input or fine increments with the up and down arrows; the two other values will change accordingly.

Please note precision of the distance value depends on correctness of the temperature value inserted in the main setup. In a concert hall with an audience present, there will probably important temperature variations, so this value should only seen as a rough measure.

Lastly, remember the delay value in samples is the master value, from which others are derived.

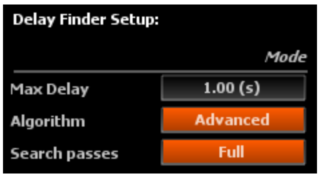
12.4.7 Find

Clicking the button starts a new delay value computation. Previous values, whether computed using the delay finder or entered by hand, will be erased. The algorithm accumulates a certain amount of incoming signal before the actual computation is actually performed, to ensure the delay is always computed using the most current audio.

12.4.8 Progress

An informational text showing the progress of the computation is shown when the delay find button is clicked, as well any error potentially encountered.

12.5 Setup



Delay finder setup options

12.5.1 Max delay

Sets the maximum delay that can be computed. The default is 1 second, which equates to a maximum distance between microphone and speakers of roughly 300 meters, and should be large enough for most practical applications. You can decrease this value as this minimizes the possibility of false readings.

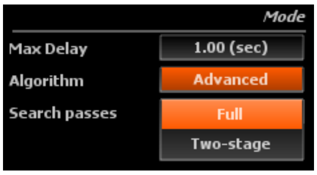
12.5.2 Algorithm

Selects between three different delay finding algorithms:

- Basic: lowest CPU load, less robust to noise and interference.
- Standard: medium CPU load, the default.
- Advanced: heavy CPU load, can help in very noisy environments.

In the rare case where the standard method fails in your particular environment, you should try other methods.

12.5.3 Search passes



The delay can be set to work in one or two passes:

- Full (default): one search pass covering all possible values.
- Two-stage: first pass to determine a rough delay value, followed by a second to refine the reading.

Two-stage delay finding can improve accuracy in the context of an environment with heavy multiple reflections.

13 Signal generator

13.1 Signal types

13.1.1 Pink noise

Pink noise is a random signal with an amplitude falloff inversely proportional to frequency. This is the most commonly employed variety noise in audio measurement, as it a constant-energy perceived content.

13.1.2 White noise

White noise is a random signal with constant energy across the audio range. Compared to pink noise, it sounds much brighter as it has more energy in high-frequencies. Commonly employed for electronic apparatus measurements.

13.1.3 Sine

Fixed-frequency, pure tone generator.

13.1.4 Sweep

Generates a variable tone from start to end frequencies. Linear and logarithmic variants are available. Log. sweep is best suited for audio measurements as this corresponds to constant time per octave.

13.2 Controls

Signal generator controls

13.2.1 Type

Sets the signal type (see p.87) to generate.

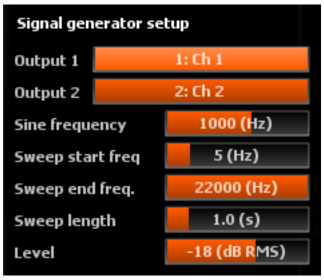
13.2.2 Level

Output level of the waveform, expressed in dB RMS.

13.2.3 Enable

Toggles signal generator output on and off.

13.3 Setup



Signal generator setup options

13.3.1 Output

Select the hardware output(s) to which the signal generator should be routed. Set to 'None' to disable the signal generator output entirely.

Output 1

First generator output.

Output 2

Second generator output.

Remarks

Both signals sent to the hardware output channels are identical.

13.3.2 Sine frequency

Sets the frequency of the sine generator, only applicable when the signal type is set to Sine.

13.3.3 Sweep start/end frequencies

Sets the range of frequencies to sweep.

Remarks

Reverse start and end frequencies to obtain reverse sweep.

13.3.4 Sweep length

Sets the overall duration of the sweep in seconds, *i.e.* the time taken to go from start to end frequency.

13.3.5 Level

Generator output level in dB RMS.

14 Transfer function measurement

14.1 Introduction

The transfer function of a system measures its frequency response, which is expressed in terms of magnitude and phase response. The transfer function measures the way the system affects the magnitude and phase of an incoming signal at different frequencies, and is essentially a ratio of output versus input spectra.

Practical uses of this are numerous: determining the curve of an equalizer, determining what frequencies are emphasized by an outboard device, measuring a room's acoustic response, etc.

Remarks

The transfer function assumes the system meets the following conditions

- linearity
- time-invariance

Linearity notably implies the system is free of distortion, and time-invariance that its characteristics do not change in time. Failing to meet this requirements will lead to unpredictable results.

In practice, the transfer function is considered an adequate measurement technique for most real-world systems, except for devices exhibiting highly non-linear behavior such as compressors and distortion effects, and time-modulation based effects such as chorus and flanger.

14.2 Transfer function magnitude

The transfer function magnitude displays the gain versus frequency curve of the system under test. A pass-through obviously results in a flat horizontal line centered on 0dB. This line represents the ideal curve one would be able to achieve if all the systems defects could be compensated for, and that serves as a reference target when doing room correction.

14.3 Transfer function coherence

Coherence is a normalized - that is comprised between zero and one - measure of the confidence of the transfer function at a specific frequency. In other words it describes how trustworthy the transfer function is at the corresponding frequency.

Coherence at a particular frequency indicates wether the system can accurately be described as linear gain and phase shift or not.

Interpretation and uses

Low coherence most often indicate a bad measurement, so you should look for possible causes and correct them before starting again.

Typical culprits include a noisy device, presence of distortion, channel crosstalk, acoustical noise such as cooling fans, people talking, handling noise, bad isolation from the outside, etc. Low coherence also manifests when delay is improperly compensated for.

While maximizing coherence is desirable, in most cases, it will most likely be impossible to attain a flat curve approaching unity at all frequencies, except in an anechoic chamber or very 'dead' sounding room with minimal reflections.

Reverberation, as well as mismatched transducers, tends to give lower coherence, as the signal arriving at the microphone position is really the sum of several time-delayed version of the source.

Sometimes it will be impossible to get good overall coherence, and the magnitude and phase curves will therefore be less precise, stable and smooth. This does not mean you cannot attempt extract any information from those. As always, use your judgment and knowledge of the specific system to decide which assumptions seem reasonable.

Display

By default, the transparency of the main magnitude curve is also modulated with the coherence values, which increases readability by effectively dimming untrustworthy curve portions. In addition to controls and settings identical to those of the spectrum magnitude curve, you can toggle the *coherence curve* on and off with the 'Enable' switch under ‘Coherence’ in the settings.

14.4 Transfer function phase

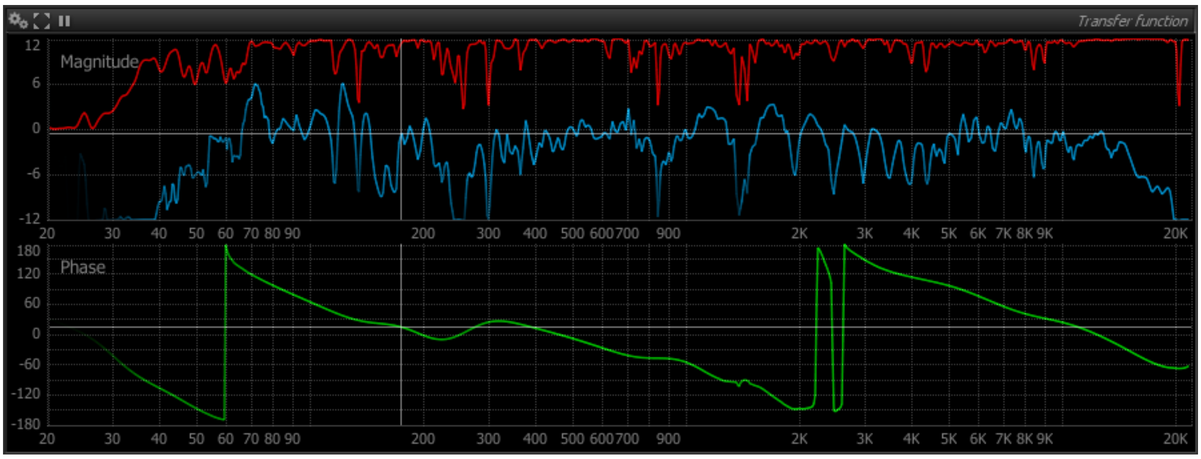
Phase information is sometimes overlooked, and indeed it is less straightforward to understand and interpret than magnitude. Altering the phase of a signal can range from subtle to dramatic, and phase distortion can lead to temporal smearing of the audio, loss of spatial information, and other nuisances.

The transfer function phase curve displays the phase difference between output and input of the system at different frequencies, in degrees, ranging from -180 to 180.

Remarks

Due to the definition of phase itself and the means of computing it, the curve is generally more sensitive to extraneous noise, distortion and time-varying conditions. Even more so than with the magnitude curve, a precisely compensated delay is critical to accurate phase computation.

In very reverberant environments, the phase curve will be very chaotic. This is inevitable and a direct consequence of the complex nature of the system, and not a limitation of the instrument. We advise to use Pure spectrum analysis mode which mitigates phase computation inaccuracies compared to plain FFT.



Typical transfer function display in a live room

Notes

Pure Analyzer employs several smoothing algorithms custom designed for phase curve smoothing, as explained in the section about Phase (see p.95) setup.

14.5 Setup

Transfer function setup options

Notes

Time averaging is on by default as the goal here is to provide the most stable display, and to eliminate any variations of the signal in time.

Frequency smoothing can be useful to smooth out irregularities and get a general picture of the curve. It is advised to use this function sparingly though, as it can change values by a large proportion, and obscure potential problems with either the actual system being measured, or the measurement setup itself.

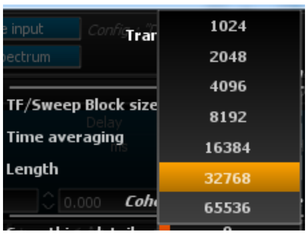
A combination of time averaging and frequency smoothing is most often required to obtain readable results in real-world scenarios, especially with large rooms and outdoors.

14.5.1 Main

TF/Sweep Block size

Block size used for the transfer function and the snapshot done with sweep. The default is 32768, which is appropriate for most cases.

Increasing this value gives better frequency resolution, at the expense of CPU load. Lower values can be employed if you're only interested in the overall response of the analyzed system.



Transfer function block size

Time averaging

Toggles time averaging on and off. Default is on, which in most cases is necessary to provide a stable display readout.

Length

This setting determines the number of blocks taken into account to compute the averaged transfer function. Increasing this value will give a smoother readout, but the display will react more slowly to any input variations, and CPU load will be higher.

The default is 32.

14.5.2 Coherence / magnitude

Smoothing detail

Sets the amount of detail present on the smoothed magnitude and coherence curves. This number is roughly the maximum number of valleys and peaks that will remain after smoothing. A low value of around 10 is good for getting a global and uncluttered picture of a room’s frequency response.

Notes

Relying on smoothed curves altogether should be avoided, as smoothing can mask-out essential information such as room modes, which materialize as sharp peaks and dips in the transfer function magnitude curve. We strongly recommend basing your judgment on both raw and smoothed curves even when the raw curve is very noisy.

14.5.3 Coherence

Enable

Toggles the display of the coherence curve on or off. With multiple snapshots, the display can rapidly become crowded, and in that case hiding the coherence curves will improve legibility. In the general case however, we recommend leaving this enabled as coherence represents important information which should not be overlooked.

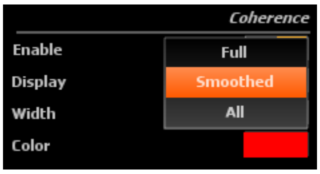
Use for curves transparency

Allow to use the coherence values to define Magnitude and Phase curves transparency.

Display

Toggles between one of three modes:

- Full : main unsmoothed coherence curve.
- Smoothed: smoothed coherence only.
- All: both.



Available coherence display modes

Width

Size of the pen used to draw the coherence curve.

Color

Color of the pen used to draw the coherence curve.

14

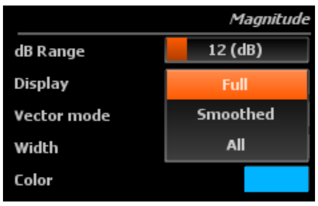
14.5.4 Magnitude

Range

Minimum and maximum values to which the display is clamped, in decibels.

Display

Toggles between various combinations of raw and smoothed magnitude curve display.



Available magnitude display modes

- Full : main unsmoothed magnitude curve.
- Smoothed: smoothed magnitude only.
- All: both.

Keep in mind the smoothing process can filters out a lot of information, so relying solely on the smoothed curve should be avoided.

Vector mode

Toggles vector averaging of the transfer function magnitude on and off.

Vector mode computes the average sum of magnitudes and magnitudes multiplied by coherence. In vector mode, the averaged magnitude is therefore and indication of the perceived magnitude spectrum, *i.e.* the sum of the direct path and diffuse field signals.

Default is off.

Auto-Range

Toggles auto-range on and off. When enabled, the display range automatically follows that of the transfer function magnitude curves, which is useful for hands-free operation, for example. Default is off.

Width

Size of the pen used to draw the magnitude curve.

Color

Color of the pen used to draw the magnitude curve.

14.5.5 Phase

Phase curve specificities

You will notice the phase curve is generally very sensitive to spurious noise and interference, and that in general it requires a bit of work on your part in order to read and interpret it. Outside of the studio, in noisy places such as a live venue, phase smoothing is almost always mandatory in order to get a readable curve. It is important to understand that smoothing destroys information in order to achieve this, so you should always double-check what you see on the smoothed curve against the original, raw data.

The algorithms employed here are specific to phase, and have more options than the regular smoothing employed for spectrum magnitude, transfer function magnitude and coherence, in case you wish to fine-tune their behavior.

Display

Toggle between the various phase curve display modes:

- Full: raw phase only.
- Smoothed: smoothed phase only.
- All: both.

Smoothing detail

Adjusts the overall level of detail that remains after smoothing, in percent. Do not set this too low or you might miss out important information such as phase shifts at critical frequencies such as those associated with loudspeaker crossover networks.

Values around 30 are appropriate in the general case.

Smoothing threshold

Amount of relative local phase variation that is allowed to pass through. Raising this filters out local phase curve detail, such as noise. Setting it to one suppresses all detail, whilst setting it to zero leaves the curve untouched.

0.60 is a good starting point.

Smoothing method

Please refer to Phase smoothing methods (see p.97).

Smoothing passes

Sets the number of smoothing algorithm iterations. You can apply the smoothing process several times in order to get better results whilst still retaining local detail. Increasing this value requires more CPU processing power, so it is advised to lower this value if you find your computer cannot cope with the load. Default is 5.

Smoothing Hide jumps

When enabled, the portion of the curve that corresponds to a phase rotation is not displayed.

Smoothing uses coherence

When enabled, frequency regions of the phase curve with low coherence are applied more smoothing. Conversely, regions with coherence close to one are applied little or no smoothing.

Low-coherence regions are caused by low signal-to-noise ratio, multiple paths, etc. which cannot be accurately described in terms of a simple gain and a phase shift anyway, so it makes sense to suppress excess detail in these regions to improve the curve's general readability.

Width

Size of the pen used to draw the phase curve.

Color

Color of the pen used to draw the phase curve.

14.5.5.1 Method

Available phase smoothing algorithms.

The general principle is that for each curve pixel, the algorithm determines the amount of smoothing applied in its neighboring region, based on a threshold determined from other pixels in the region. The smoothing therefore adapts to the curve content, applying more smoothing in noisy regions.

StdAvg/Abs was determined to be method giving the best results in the general case, and is set as default. You might still want to experiment with other algorithms, especially if you have a slow computer.

Fix/Abs

This is the simplest and least CPU-intensive phase smoothing algorithm. Smoothing uses surrounding pixels below an absolute threshold.

In practice, this means curve regions with large variations are applied stronger smoothing.

Fix/Rel

Same as above, using a relative threshold.

Var/Abs

A variant of first algorithm.

Std/Abs

The threshold is determined from the pixels standard deviation, which is a statistical measure of data variation.

Std/Rel

Same as above, using a relative threshold.

StdAvg/Abs

Combination of above methods, using absolute threshold.

StdAvg/Rel

Combination of above methods, using relative threshold.

Notes

Please keep in mind the smoothing process is purely a visual aid, and is not intended to compensate for an inadequate measurement setup. In short: always rely on your ears and scientific knowledge first !

14.5.6 Other

Color grading

Apply frequency-dependent coloring to the curve. Default is off.

Zoom

Curve zoom ratio slider.

14

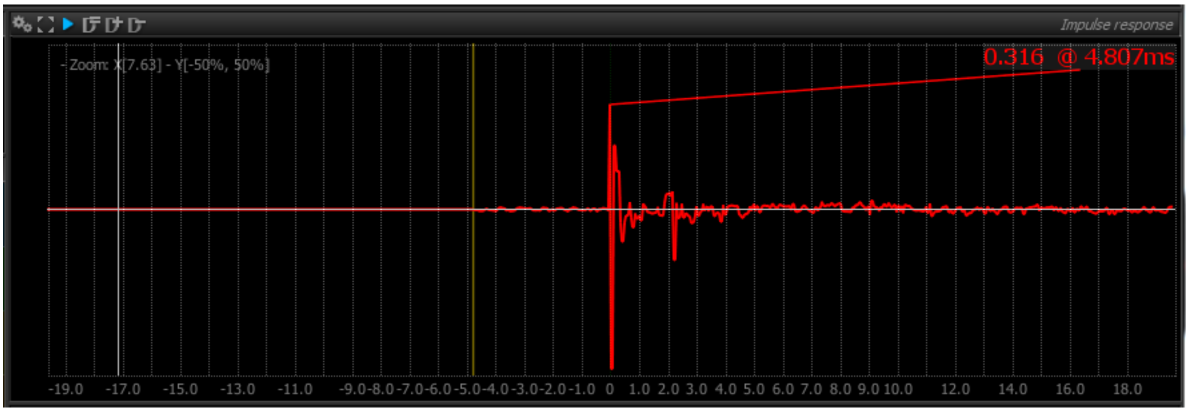
15 Impulse response measurement

15.1 Introduction

The impulse response of a system is the signal obtained at the output when feeding a click (alsto termed impulse, spike or Dirac) its input. It is a fundamental tool to describe the time properties of a linear system.

Combined with the transfer function, impulse response measurement is essential in characterizing the acoustics of a studio, concert hall or venue, from which synthetic figures such as reverberation time are derived. Determining the impulse response of an amplifier and loudspeaker in tandem can also serve to assess their performance.

A pass-trough device, or equivalently, a completely dead space such as an anechoic chamber exhibit a unit impulse response, whose value at zero time is gain, and is zero at all other instants.



Impulse response display example

Analyze / freeze

The button toggles the impulse response real-time update on and off.

Delay Set

The delay Set button set value of the peak time location to the delay value currently set for microphone channels in the Live IO (see p.82) panel.

If Real Time curve is disable, the Max value of the selected snap shot is used.

Delay add

The delay add button adds value of the peak time location to the delay value currently set for microphone channels in the Live IO (see p.82) panel.

If Real Time curve is disable, the Max value of the selected snap shot is used.

Delay subtract

The delay subtract button subtracts the peak value to the microphone channels delay.

If Real Time curve is disable, the Max value of the selected snap shot is used.

Notes

The impulse response is closely tied to the transfer function, in that they are both related to another by a Fourier transform.

For practical aspects, Pure Analyzer employs two distinct analysis engines to compute the impulse response and transfer function, as this allows to use separate settings for the two, which is often necessary in practice.

15.2 General procedure

Impulse response (IR) measurement requires that sufficient samples be accumulated before the actual computation is ready, depending on the values of the Max Length (see p.101) and Time averaging (see p.102) settings. The user interface displays a message indicating the remaining time before the display is ready, whenever the related settings are changed or the reset button is pressed.

Because the software cannot detect whenever you make changes to the analyzed system, you need to press the Reset button in the setup or wait for the display to stabilize before reading the display.

Once your test setup is ready, press the ‘Reset’ button and wait for the display showing remaining time to disappear, at which point the IR display is ready. When a sufficient amount of samples have been accumulated, IR computation goes on as long as the ‘Run’ button is active, and is updated with new incoming samples.

Notes

Make sure the actual impulse response is shorter than the maximum specified time, otherwise mild to severe time-aliasing will occur, and the measurement will not be reliable. A good rule of thumb is to set the Max length parameter to twice that of the estimated RT60 of the room.

If in doubt, raise the Max length setting until the impulse response curve does not change, and check the tail of the curve does indeed fall to zero.

15.3 Time averaging

The time averaging function computes the mean of several IR measurements over time, which is very useful to filter out noise and other artifacts. It is enabled by default as this gives better display stability and measurement robustness, however averaging also slows down the reactivity of the display to incoming variations, so you can disable it if needed.

When IR averaging is enabled, a message is shown giving the number of currently computed impulse responses versus averaging length. The display switches to show the mean confidence percentage when ready.

15.4 Main setup

Impulse response setup options

15.4.1 Run

Toggles impulse response live update on and off. Default is on. You can temporarily freeze the impulse response with this button, to examine it in detail at your leisure, without worrying about changing external conditions.

Disabling ‘Run’ is equivalent to freezing the measurement, and leaves the averaging buffer as is.

15.4.2 Reset

Resets the impulse response computation, including the averaging buffer.

Notes

If you are using a lengthy averaging setting and have just have changed your setup, you can reset the entire impulse response to immediately forget previous measurements .

15

15.5 Time

15.5.1 Max length

Sets the maximum length of the impulse response in seconds. If the reverberation time in your room exceeds this value, time-aliasing will occur, meaning that the impulse response computation will be incorrect and some of the reverberation tail might end up at the start of the display. The default value is 0.3s.

Increasing this value not only requires more processing power, it also increases the time needed to wait for the display to be updated, as the calculations involved need a greater amount of incoming audio samples to be processed.

Combining time averaging and a long length setting mean you'll have to wait 30 seconds or so for the display to stabilize, so you should really do this if you need to or do not mind waiting.

15.5.2 Time averaging

Accumulates several impulse response measurements and averages them before display. This allows for more precise measurements and lessens the effect of spurious acoustic noise interfering with the measurement, but it also means having to wait longer for the measurement to be ready.

15.6 Scale

15.6.1 AutoRange

Toggles auto-scaling the vertical axis to the effective range of the impulse response data in the current timeframe. It functions as an automatic zoom alongside the vertical axis, which can provide useful for hands-free operation.

15.7 Other

15.7.1 Zoom

Zoom X

Adjust the horizontal axis zoom factor, which can also be changed by clicking inside the impulse response display itself and rotating the mouse center wheel up and down, if your mouse has this feature.

Zoom Y-/+

Adjusts the vertical axis zoom factor.

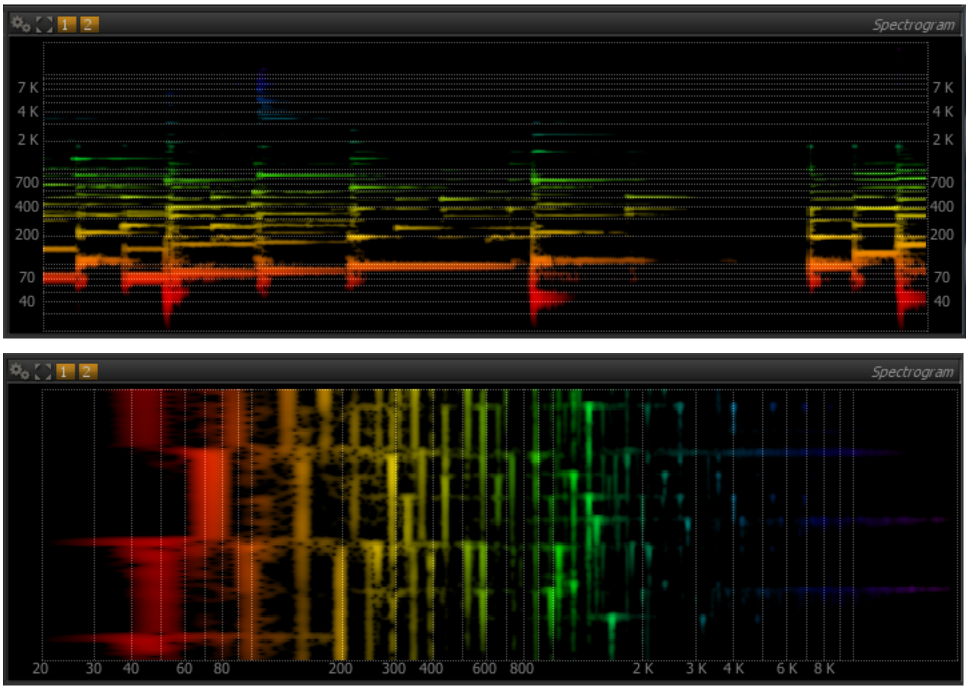
16 Spectrogram

16.1 Usage

The spectrogram is a two-dimensional view of the evolution of the signal's spectrum over time, i.e. a frequency (Y-axis) versus time (X-axis) plot (or the invert; depending on the direction setting), with the magnitude modulating the color and intensity of the pixels.

A spectrogram can be computed using the STFT (short-term Fourier transform) as well as other means. It serves as a useful tool to get a global picture of how the frequency content of a signal changes over a time, and eases identification of its structure. Broadband noise appears as background, a pure tone tone as a horizontal line, and a transient as a vertical line.

Harmonic content appears as horizontal groups of parallel lines and vertical bars respectively, etc.

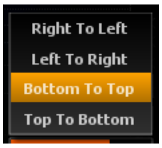


Example spectrogram view

16.2 Setup

Spectrogram setup

16.2.1 Direction



Define the scrolling direction of the spectrogram.

16.2.2 Log Gain

Toggles logarithmic scaling of the magnitude spectrum on and off.

Default is on.

When enabled, the magnitude at a time-frequency is applied a logarithmic scaling before being converted to a pixel value. This has the effect of compressing the dynamic range, and makes low energy components stand out more, but also decreases the contrast of the display.

16.2.3 Threshold

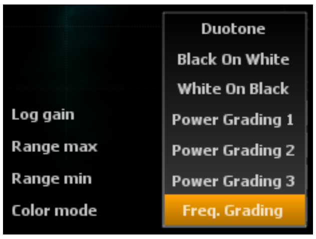
Threshold min

Sets the minimum amplitude spectrum value to be displayed.

Threshold max

Sets the maximum amplitude spectrum value to be displayed.

16.2.4 Color Mode



Duotone

In this color mode, the amplitude of a time-frequency point is mapped to a pixel using a two-color palette, set using start/end colors.

Black On White

In this color mode, the amplitude of a time-frequency point is mapped to a pixel using a Black & White color palette with White as background.

White On Black

In this color mode, the amplitude of a time-frequency point is mapped to a pixel using a Black & White color palette with Black as background.

Power grading 1, 2, 3

In this color mode, the amplitude of a time-frequency point is mapped to a pixel using different predefined color palette.

Frequency grading

In this color mode, the amplitude of a time-frequency point determines the intensity of the corresponding pixel, whose color varies according to frequency.

Duotone start/end colors

Sets the color to use for minimum and maximum amplitude components respectively, when color mode is set to 'Duotone'.

17 Snapshots

Curves can be saved on disk for subsequent loading, allowing for comparison between mixes, comparison to a reference spectra, etc.

A snapshot contains the state of the curves at the time it was taken:

- Channel spectra.
- Transfer function.
- Impulse response.

A snapshot, as implied by the name, is like a picture of the whole application at a given time. A snapshot contains all the data to save the current signal analysis as displayed on screen, and restore it at any given time, as well as to make comparisons between different locations, setups, etc.

17.1 Usage

Snapshots

Any number of snapshots can be stored and recalled for further use, and are organized into a group container called a project. Please keep in mind computing and displaying the data associated with a snapshot is not free in terms of processing power and memory. How many snapshots you can use at a time will depend on your particular configuration.

Project

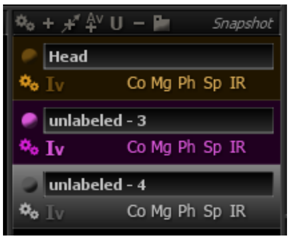
Pure Analyzer creates a default project at startup, which the snapshots will be added to. Projects are stored on a disk as a folder containing associated data files. Projects can therefore be renamed, moved, archived and transferred between computers using any method you wish, provided you include all data files inside the project folder.

You can save and reload as many projects as you want, disk space permitting.

Notes

Projects are saved in `<User folder>/Flux/PureAnalyzerSystem/<Project Name>`.

17.2 Controls



Snapshot list and controls

The snapshot area shows a list view, where one or more snapshots can be selected. The selected snapshot(s) will be highlighted accordingly, both in the list and the corresponding display(s), with increased curve thickness .

17.2.1 Selection and navigation

The snapshot list follows standard user interface guidelines, which means you can:

- Use keyboard up and down arrows to change the currently selected snapshot. Note: the snapshot area must have focus for this to have effect.
- Click on any snapshot to select it.
- Shift-click to define a selection range of multiple snapshots.

17.2.2 Add new snapshot

Clicking the icon immediately creates a new snapshot, stores it on disk, adds it to the current project and selects it.

17.2.3 Acquire sweep

The button launches acquisition of a sweep snapshot. This special type of snapshot automates the acquisition of transfer function and impulse response curves using a swept sine generator output.

Please check the following for proper operation:

- Generator output (see p.88)(s) should be properly assigned to the corresponding hardware channels
- Hardware IO (see p.21) should be properly configured and set to hardware output(s)
- Sweep start/end frequencies (see p.89) should be set as desired

Providing the previous requirements are met, a progress dialog will then be displayed until all data has been acquired and the snapshot is computed and ready for display.

Notes

Ensure the outputs of the generator and the connected speakers are set to reasonable levels in order to prevent damage to your equipment and hearing loss.

17.2.4 Create average

Click the button with multiple snapshots selected to create a new snapshot average of these.

The new snapshot curve data is computed from the selected snapshot data as follows:

- Spectrum magnitude: average of magnitude vectors.
- Transfer function magnitude: average of magnitude vectors.
- Transfer function phase is set to zero as there is no mathematically significant meaning to averaging of potentially unrelated phase spectra.
- Transfer function coherence: average of coherence vectors.

- Impulse response: average of signals.

The averaging can only be performed if the snapshots are compatible with one another, that is they have identical:

- Sampling rate.
- Number of channels.
- Spectrum type.
- Impulse response length.

A warning message will inform you the averaging cannot be performed if one of the above conditions are not met.

Remarks

The snapshot average stores the average of the snapshots at the moment it was created. If you change the snapshots in any way, the snapshot average will not change.

17.2.5 Update current

Clicking the U button will overwrite the last selected snapshot contents with the most current data.

This is especially useful when you are fine-tuning your measurement setup and only want to keep the latest one, without creating several snapshots and deleting them afterwards.

Notes

This function is destructive: there is no means to revert the original snapshot data.

17.2.6 Load project

Opens a dialog box where you can select an existing folder containing a previously saved project.

To create a new empty project, creating a new folder and name it, then selecting using in this dialog.

17.2.7 Curve visibility

For each snapshot, you can control which curve should be displayed. These controls are intended to select only those curves that you really need to be displayed when there are many visible snapshots, and still maintain a legible display:

- Transfer function coherence.
- Transfer function magnitude.
- Transfer function phase.
- Magnitude spectrum.
- Impulse response.

Notes

The default visibility of newly created curves can be customized in Display defaults (see p.109).

17.2.8 Color

Opens up a color selector dialog where you can manually set the color used to identify the snapshot, both in the list and as a curve.

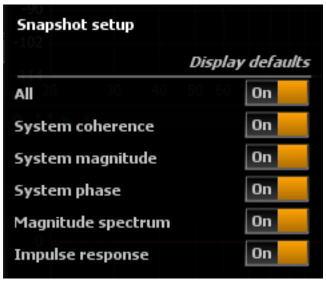
17.2.9 Name

By the default, newly created snapshots are given the name `unlabeled-x`, where `x` is the current number of snapshots in the project. You are strongly encouraged to edit this name for further reference.

17.2.10 Invert (Iv)

Invert the magnitude curve of the Transfer function

17.3 Setup



Snapshot setup options

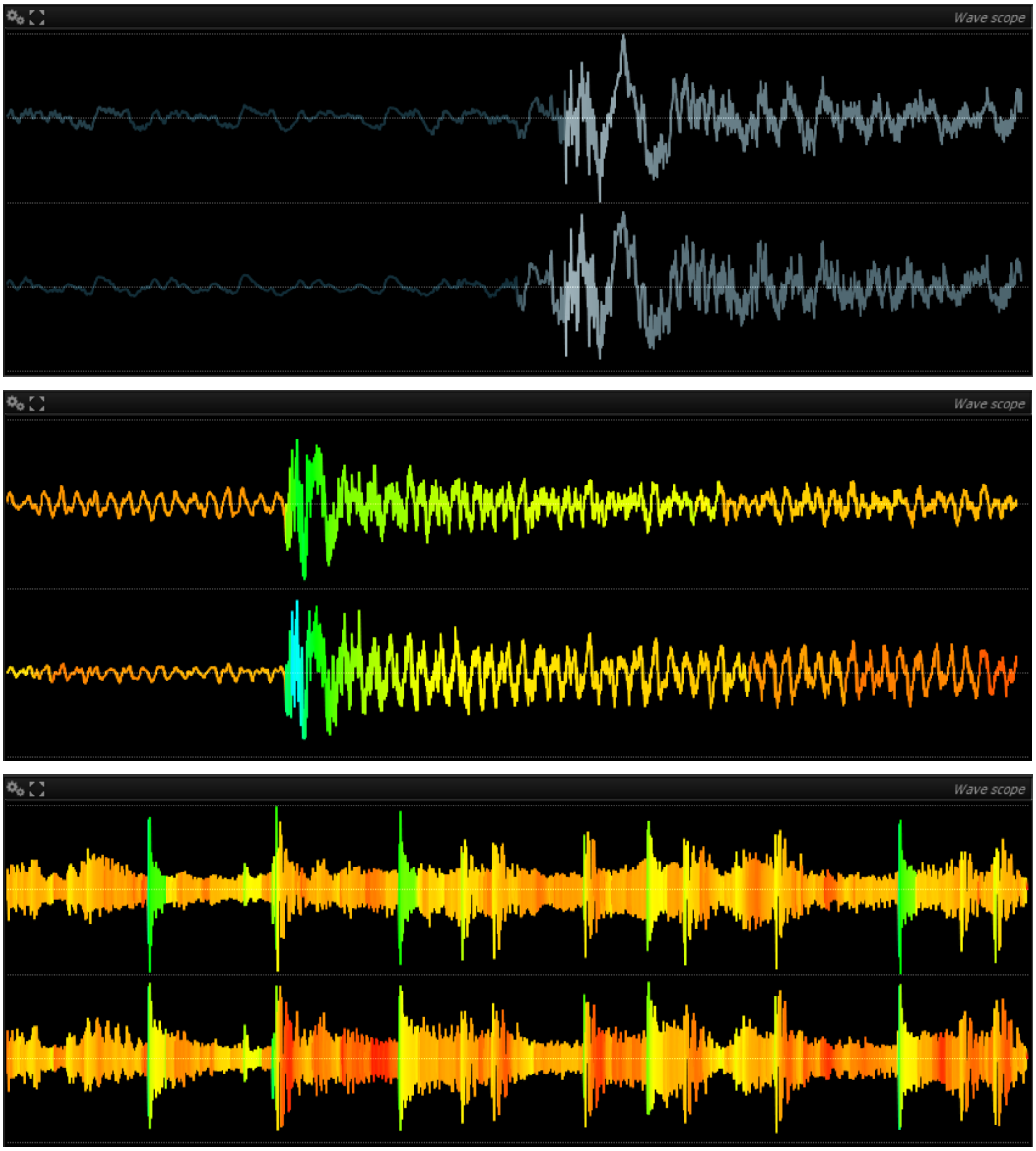
17.3.1 Display defaults

Toggles the default curve visibility applied to newly created snapshots.

'All' controls whether new snapshots should be visible by default, and you can fine-tune which curves should be shown/hidden here also.

18 Wave scope

The wave scope is a simple oscilloscope-type waveform display.



Wave scope display with stereo input

Notes

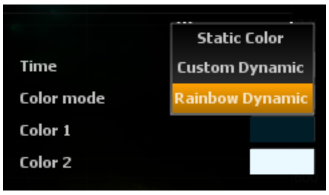
The wave scope will include more functionality and settings in future releases.

18.1 Setup

18.1.1 Time

time window in millisecond

18.1.2 Color Mode



Static

Display the waves using 1 unique static color.

Custom Dynamic

Display the waves according to the transient using a 2 user defined colors gradient .

Rainbow Dynamic

Display the waves according to the transient using a rainbow colors gradient .

19 System requirements

Pure Analyzer is built around Flux::'s new 2D/3D efficient graphic engine, which employs full GPU-acceleration using an OpenGL-compliant graphics card.

In order to experience the outstanding responsiveness with Pure Analyzer, even with a very busy display, and to fully take advantage of the software's analysis capabilities, using a modern nVidia or ATI Radeon graphics card is recommended.

Older, and other less efficient graphics cards do not have the required performance and specifications, and offload too much work to the CPU (see below).

The processor is also an important factor, and we recommend using at least and Intel Core 2 Duo, Core i5 or newer architecture processor. AMD processors are also supported, but might exhibit lower performance, as they do not offer the same capabilities and optimizations as Intel CPUs.

19.1 System recommendations

Minimum requirements

- CPU: Intel Core 2 Duo.
- GPU: OpenGL 2.0 or superior compatible, with pixel-shader support.

Recommended configuration

CPU: Intel Core i5 or better.

GPU: AMD/ATI Radeon or nVidia video-card. Intel integrated graphics are not powerful enough and should be avoided.

Common requirements

A free USB port to connect the Flux:: or iLok dongle.

Notes

Please check the latest version of vendor-provided, optimized drivers are installed for your graphics card. Generic drivers are generally less up-to-date and may contain bugs or miss optimizations present in drivers specific to your particular model.

19.2 Compatibility

Pure Analyzer is a 32bit application fully compatible with 32 and 64-bit operating systems.

Operating Systems

- PC: Windows - XP, Vista, 7.
- Apple: Mac OS X versions 10.5, 10.6 and 10.7.

Hardware IO support

Any soundcard with a driver compliant with these standards:

- ASIO(Windows).

- Core Audio (Mac OS X).

SampleGrabber is a 32-bit plug-in compatible using 64-bit double precision internal processing, compatible with 32-bit and 64-bit (via bridge) hosts

All major native formats (VST, AU, RTAS) and TDM (Avid Venue D-Show compatible)* are supported.

Supported formats

- Windows - XP, Vista, 7
  - VST (2.4)
  - RTAS*
  - TDM*
- Mac OS X - 10.5, 10.6, 10.7
  - VST (2.4)
  - AU
  - RTAS*
  - TDM*

Notes

*The TDM/RTAS version requires ProTools version 7 or above.

20 Credits

Project manager and Designer	Gael Martinet
Developed by	Gael Martinet Siegfried Hand Lorcan Mc Donagh Samuel Tracol
DSP Algorithms Specialist	Lorcan Mc Donagh
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and thanks to all fantastic testers...

Special Thanks to Alain, Yves, Bruno, Cyril and Claude for helping to shape our minds over the years.

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