Track Range

Contents

1 Introduction

1.1 System Requirements

1.1.1 Windows

The Track Range Plugins work with Windows XP, Vista, 7, 8, and 10, 32bit and 64bit.
VST, VST3 and AAX are supported in both 32bit and 64bit hosts.
RTAS is supported in 32bit ProTools 7 or newer.

1.1.2 Mac OS

The Track Range Plugins require OS X 10.5 or newer on Intel Mac.
VST, VST3, AU and AAX are supported in both 32bit and 64bit hosts.
RTAS is supported in 32bit ProTools 7 or newer.

1.2 Installation and Setup

1.2.1 How to Install on Windows

You need two files to install; the plugin installer and the license file. Both are available from the Downloads area of the DMGAudio site whilst logged in. If you're using a demo, download the installer from the site, and use the form to have your license file emailed to you.
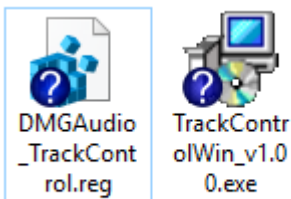
Double-click license.reg, and this box will open:
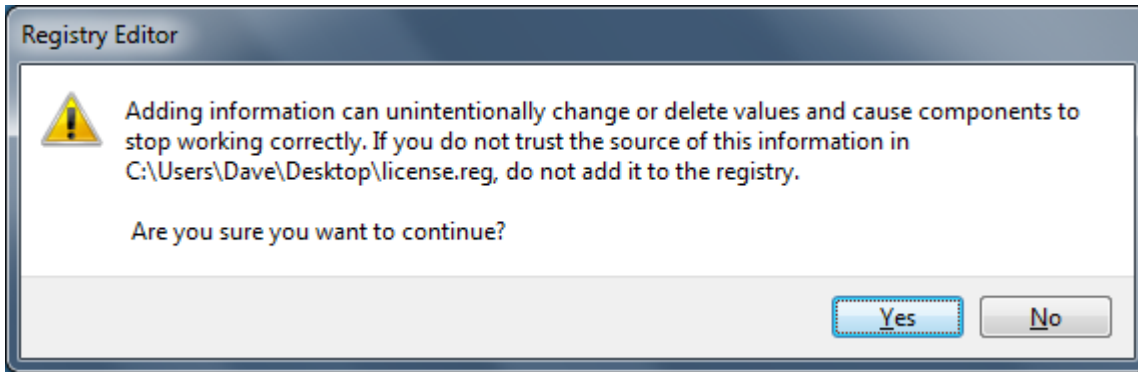
Click "Yes" and the license is installed.

Now double-click the zip-file which contains the installer.

Double-click the installer; it will look like this:
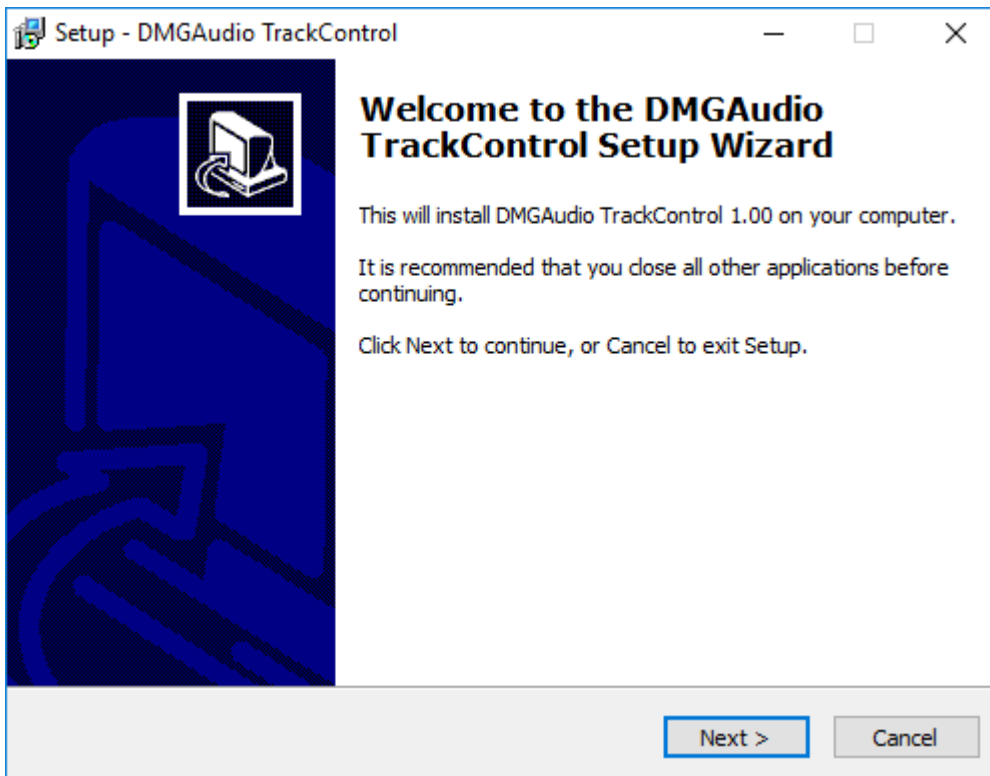
Click "next" (changing any options if you need to), and your Track Plugin is installed!

Reload your host, and enjoy!

1.2.2   How to Install on OS X

You need two files to install; the plugin installer and the license file. Both are available from the Downloads area of the DMGAudio site whilst logged in. If you're using a demo, download the installer from the site, and use the form to have your license file emailed to you.
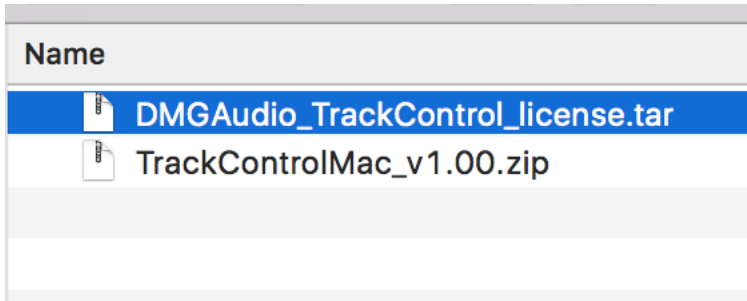
If you are using Safari, one or both of these files may be automatically unzipped for you. If not, double-click them to decompress them, so you get this:
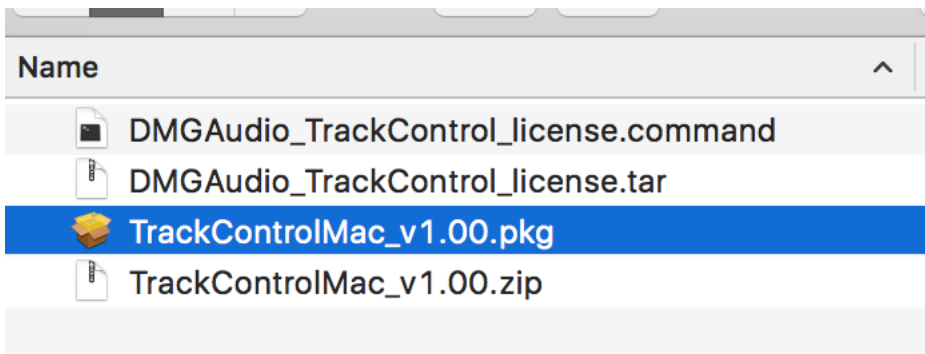
Double-click "license.command" to install the license. The license will install, and you'll get the following message to confirm:
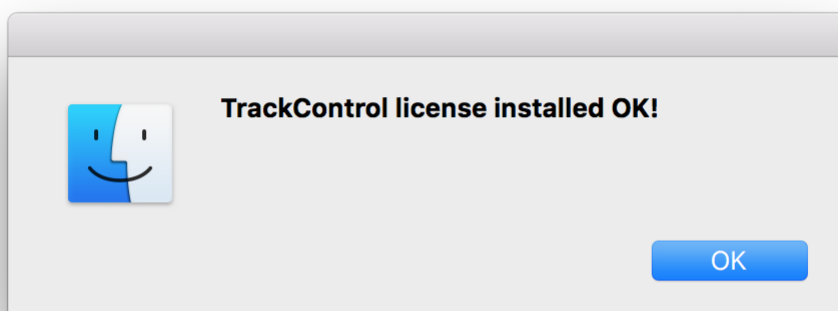
Now double-click the installer, and it will launch. It looks like this:
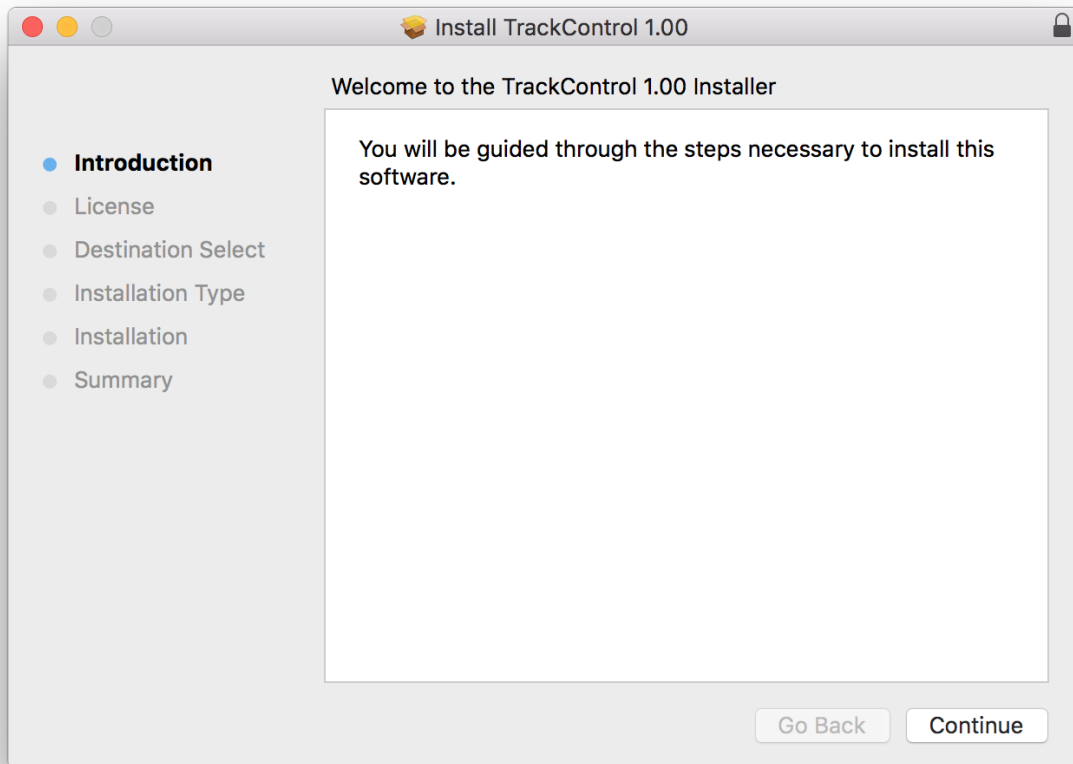
Click through until it installs, and everything is set! Now relaunch your host and enjoy!

2   Getting Started With The Track Range

The Track Range is your one-stop upgrade for all the tools you use non-stop while mixing.

They make problem-solving faster, and creative processing more immediate.

With each plugin we've tried to build something that's best-in-class for mix tools, whilst keeping CPU and Latency low, and a sleek interface.

All too often, mix tools can end up inferior to what was around on a large-format console. Our intention is to change that, and make the jump to working inside the box that much easier.

We really hope you'll find mixing faster and more productive with these tools.

3   TrackControl

TrackControl is a general-purpose utility for making life easier in a session. Once upon a time, we'd have a large format console with a bunch of controls at the top and bottom of each strip.

Nowadays in a DAW, you can often find yourself reaching for several different plugins to replace that standard edge-of-channel functionality.

3.1   Controls

Trim and Gain are simply gain controls, provided to give you a good deal of range and fine adjustment over channel level. In general workflow, Trim is used to get the input to a workable level, and Gain is used as a high-resolution channel fader.

MS to LR will convert an MS-encoded signal (i.e. from an MS mic pair) to LR. Alternatively you can convert LR to MS using this control. This can be especially useful when using processors that do not include their own MS matrix encode/decode functionality.

Swap LR simply swaps the Left and Right channels of the input signal to flip the stereo image of the input signal.

Phase allows phase inversion (polarity reversal) of the Left, Right or Both channels. Clicking L or R will invert the appropriate channel, or clicking Phase will invert both. This is especially useful for scenarios where the distance between a source and two mics is such that the mixed signals are cancelling in an undesirable way.

HPF is a high-pass filter applied to both channels. You can set a default frequency (80Hz and 120Hz are common defaults for large-format console users), and the slope of the filter in dB/oct (18dB/Oct is our recommendation, but the range allows as gentle as 6dB/Oct, or as extreme as 48dB/Oct). Clicking the HPF label will enable or disable the filter for quick usage. It's a good working practice to keep this enabled on every non-bass channel (i.e. everything but Kick, Bass and LFE), with TrackLimit at the top of the channel.

Balance is similar in concept to a panner, in that it has the effect of moving the source from left to right in the stereo field. Balance is achieved by reducing the gain of one channel. Used in conjunction with the Width control this can allow you to fix problematic recordings by selecting one of the channels, and then pulling that back towards the middle to pan later.

Width sets the stereo width of the signal. At 0% the signal is mono. At 100% the signal is unaffected. At higher values, the effective stereo width is increased, which can enhance the perceived width of the stereo field.

Pan is used to set the stereo position of each channel. ProTools users will be familiar with this style of panner. This can be used to swap Left and Right signals, or to place a signal within the stereo field. Holding Alt while dragging will move both channels towards or away from one another.
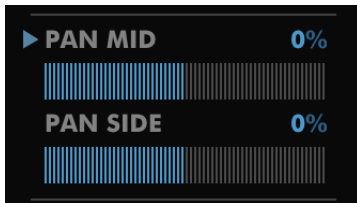
Clicking the triangle in the top-left of the Pan section allows you to select MS Pan. This allows you to position the mono content of the signal whilst leaving the stereo field content unaffected, or to move the stereo field content independently. This is a powerful technique that allows for interesting stereo effects.

Delay is used to delay a channel backwards (or forwards!) in time to compensate for recording delays. As an example, the time-delay incurred by the distance from kit to overheads can be compensated using a negative delay to bring the overhead content forward in time.

3.1.1 Mono

When inserted on a mono channel, TrackControl displays a limited set of controls: Trim, Phase, HPF, Delay and Gain are retained.

3.1.2 Output selection

The Output selection control on the menu bar allows you to audition individual components of the signal, enabling you to monitor Mid, Side, Left or Right channels.

4   TrackDS

TrackDS is a state-of-the-art De-Esser for taming overly dynamic high frequency content, such as vocal sibilance, or aggressive cymbals.

In general, you instantiate it, set the Noise Detect to somewhere around 50%, and close the plugin, having invisibly removed the offending content.

4.1   Controls

Input and Output are gains provided to make integration into your signal chain easier.

Thresh sets the threshold above which TrackDS begins to act to reduce unwanted signal level.

Style allows you to select the algorithm used for De-Essing. The "DMG" algorithm is a state-of-the-art low-latency de-esser, which should be sufficient to solve all your de-essing needs. The "Zero Latency" algorithm is provided for scenarios in which zero latency is required; it retains as much of the DMG characteristic as possible whilst functioning without latency. The "Soft Knee" algorithm is a gentler version of the "DMG" algorithm, for use cases where great subtlety is

needed. The "Exciter" algorithm inverts the gain reduction to increase high frequency content instead of reducing it; this is used to brighten dull signals (the Exciter algorithm is capable of massive gain increase, so it is suggested that you use it with small Ratio and Ceiling values).

Ratio acts just like a compressor's ratio. The bigger the number, the more aggressive the reduction will be.

Attack and Release control the dynamic behaviour of the De-Esser. The Attack time determines how quickly TrackDS acts to reduce the level. The Release guides the fully adaptive Release mechanism, which intelligently differentiates transients from harmonics to keep the de-essing effect sounding natural.

Frequency determines the frequency above which the gain is reduced. TrackDS uses a 4th order minimum-phase Linkwitz Riley crossover, which provides exact reconstruction of the signal. When the Frequency control is set below 20Hz, the crossover is disabled, and TrackDS can be used as a very subtle wideband compressor.

Noise Detect allows for automatic De-Essing. Set to 50% or higher, TrackDS will automatically distinguish sibilance and noise from signal and attenuate for you. When in use, you can set a low Threshold value, and just leave TrackDS to work for you!

Sidechain controls allow you to customise the signal to increase the specificity of the de-essing process. On individual signals, this is typically unnecessary, but in cases where you have mixed signals, and only wish to respond to one component, this can allow you to set TrackDS to target a specific signal. When adjusting Sidechain controls, TrackDS will audition the filtered signal to make setup easier.

Ceil sets the maximum attenuation that will be applied to the high frequency content. This can be used to prevent excessive gain reduction should the de-essing process become audible.

Monitoring. The dropdown menu on the menu bar to the immediate left of the Bypass button allows you to configure what signal TrackDS outputs. By default, the mode is Auto, which will give you the de-essed signal until you adjust Sidechain controls, when it will preview that signal. In a live context, that would be disadvantageous, so you can lock to the Output signal (the de-essed signal) or specifically select the S/C (Sidechain) signal. Additionally, you can monitor the Diff (difference) signal, which is the signal being removed from the input - this allows you to fine-adjust TrackDS settings, and closely monitor its processing.

5 TrackGate
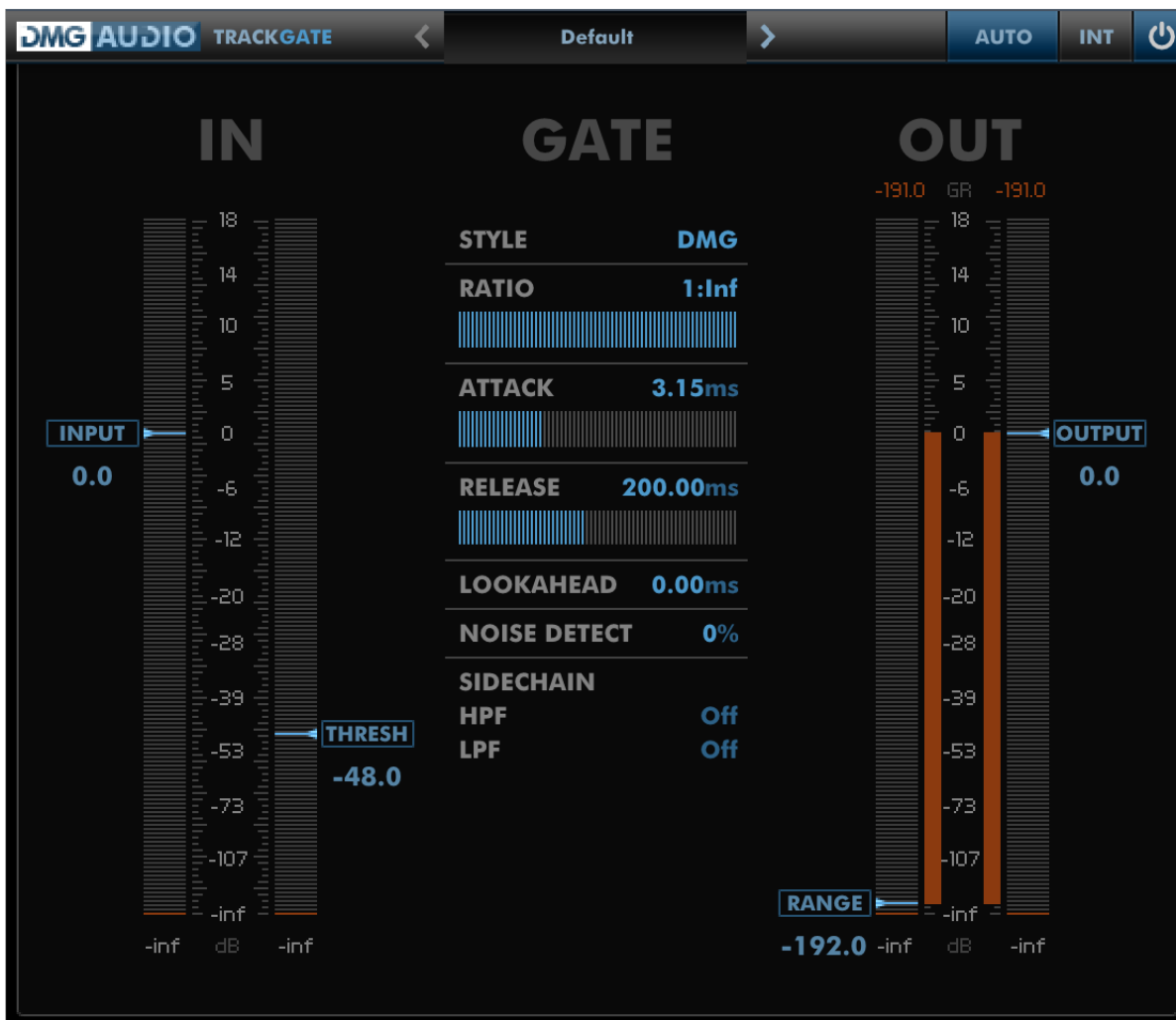
TrackGate is a fully-featured noise gate, expander, ducker and MIDI gate. A good gate can make the difference between re-recording and a great mix.

Large-format console users have had the luxury of a decent gate on-hand for years, but in the box things didn't necessarily sound quite as good.

So we applied our same obsession with circuit modelling to gates and expanders, and the result is TrackGate.

General usage consists of adjusting the Threshold until the triggering is as desired, and perhaps trimming the Ratio if expansion is preferable to gating.

5.1 Controls

Input and Output are gains provided to make integration into your signal chain easier. Note that the Input gain affects both the input signal AND the external audio sidechain input, meaning that it can be used to compensate for low-level triggers, with appropriate output compensation.

Thresh sets the level below which the gate/expander begins to attenuate level.

Style selects the gate algorithm used. The "DMG" algorithm is a tuned, precision modern gate that should be make most gating problems trivial; it features a 20ms hold time and a 6dB hysteresis. The "Textbook" algorithm has no hold or hysteresis, and allows for very short attack and release times, making it perfect for MIDI gating applications. The "Ducker" algorithm will attenuate level when the signal exceeds the threshold (rather than falls below); this is used with external sidechain input to duck background music below voiceover, or to push one element down in the mix to give another priority. The "Old Digital" algorithm is a very classic digital design. It has unusual Hold and Knee characteristics. "Brit Channel" is a circuit model of the most widely used console gate in history and is provided for large-format console users who are familiar with it. "Brit Mega" is a later British console circuit, perhaps more familiar to post-production users; it has a very similar character to the Brit Channel model, but an overall gentler effect.

Ratio sets the gating/expansion ratio. At 1:Inf, TrackGate will act as a Gate. At lower values, TrackGate will act as an expander, allowing a gentler effect. For the circuit modelled Styles, the switch is instantaneous below 1:Inf, so you'll hear different characteristics between ratios of 1:Inf and 1:500.

Attack and Release sets the dynamic times for TrackGate. The circuit models have been expanded to allow configurable ranges here. The Ducker and Textbook models allow selection of much shorter times, which are not generally appropriate for gating work, but are invaluable for MIDI effects.

Lookahead sets the lookahead time around the gating circuitry. For exact circuit model behaviour or for zero latency, set this to zero. However, for general gating work, where some latency is acceptable, small lookahead times can greatly enhance the smoothness of the gate triggering.

Noise Detect allows TrackGate to intelligently separate noise from signal, so that the gating is not triggered during passages of signal. At around 50% you have automatic gating, and can keep the Threshold very low, whilst still having it remove breaths and noise from the signal.

Sidechain allows you to filter the signal that will trigger the gate using 12dB/Oct high-pass and low-pass filters. By default, adjusting these controls will audition the sidechain signal, to simplify setting the values.

Range determines how much attenuation the gate can provide. 192dB is approximately equivalent to complete attenuation of a 32-bit signal.

Monitoring. The leftmost of the top-right menu-bar controls allows you to select the output signal. Output will always give the gated signal; Auto will automatically switch to previewing the sidechain when the filters are being adjusted; S/C (Sidechain) will output the sidechain signal.

Sidechain Source. The dropdown control between the Monitoring selector and Bypass buttons allows you to select which signal will be used to trigger the gate, between the Input (Internal)

signal, External sidechain, or MIDI. Consult your DAW documentation for information on routing of external sidechain signals and MIDI signals.

6 TrackLimit
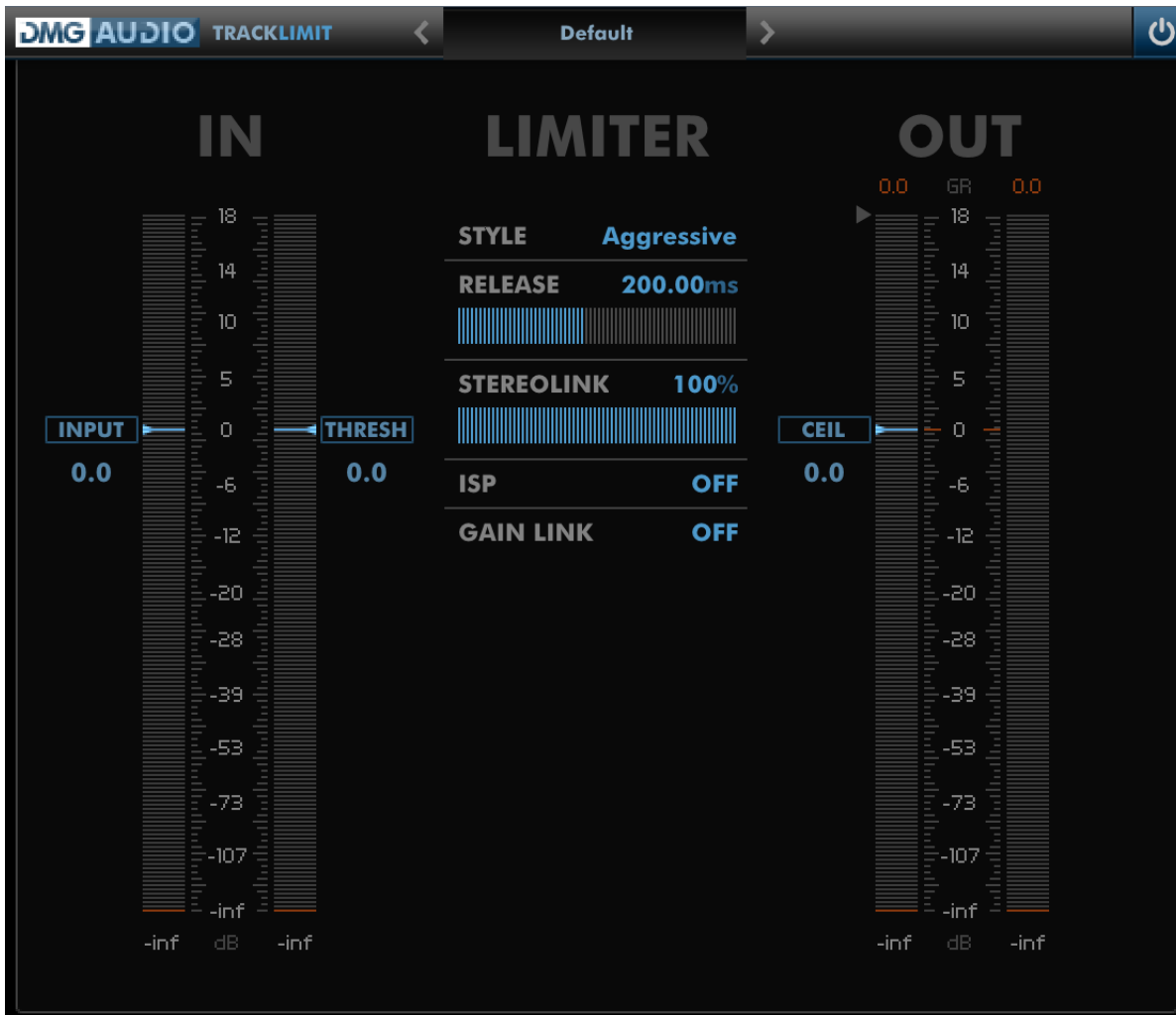
TrackLimit is a very low-latency wideband limiter, integrating the adaptive dual-stage mechanism from our flagship Limitless.

You can use it to transparently snip a few dB of peak of a channel that's triggering a bus compressor when it oughtn't, or you can use it to render out an awesome-sounding premaster.

General usage is to adjust the Threshold until desired loudness is achieved, and consider adjusting Style.

6.1 Controls

Input is a gain control to allow you to compensate for input level.

Thresh sets the amount of gain that the signal will be boosted by before limiting. The lower the Threshold, the more limiting will be applied.

Style allows you to select from a set of characters, which internally determine lookahead times, knee, release characteristics, and transient/sustain behaviour.

Release allows you to set the release time to be used when in sustained periods of limiting. The release time for transients will be much shorter, and will be intelligently determined with respect to your choice of Style.

StereoLink allows you to determine whether both channels are limited together, or whether TrackLimit can adjust each channel independently. At 100% both channels will be linked completely and the stereo image will be unaffected. At 0% both channels will be limited independently, and stereo shifting may be audible.

ISP allows you to specify that TrackLimit should use ITU1770 True Peak measurement for its loudness target, instead of dBFS. This allows you to use TrackLimit to achieve True Peak deliverable specifications.

Gain Link forms part of the traditional limiting workflow, whereby Threshold and Ceiling are linked. Enabling Gain Link will snap Ceiling to match the Threshold, so that you can audition the limiting behaviour without the gain increase. Once Threshold is set to your satisfaction, unlink Gain Link and set Ceiling to your desired target.

Ceil provides an output gain, which is used to specify the maximum level that TrackLimit can output. Typically this is -0.1dBFS for CD masters, or is provided by a deliverable specification for post-production usage.

7   TrackMeter

TrackMeter is a versatile metering plugin with a variety of metering and graph types included, in a freely resizable UI.

7.1   Controls

TrackMeter reserves almost the entirety of the user interface to present graphs.

The user interface is freely resizable, by dragging the bottom-right of the UI.

The right hand side of the menu bar offers controls to pause/resume, reset and screenshot the current graph. There's also a button to enter full-screen display, which can also be achieved by double-clicking the graph itself. Selecting Prefs opens a tray of preferences to configure the current graph display.

Below the menu bar is the graph type and source bar. Use the menu or arrows to select a graph type. At the far-right is the signal select, which allows you to select between MS and LR for stereo graph modes, or Left, Right, Mid and Side for mono graph modes (Spectrographs and Tuner)

Shift-clicking the graph acts as a shortcut for screenshots. Screenshots will be saved to your desktop, under the name "TrackMeter Screenshot <number>.png", and will match the current size of the graph.

Control/Apple-clicking the graph acts as a shortcut for pause/resume.

Many graphs respond to the mousewheel and being dragged with the mouse.

Right-clicking in the Prefs box opens a menu to allow quick propagation of settings between graph types.

Copy All Settings and **Paste All Settings** allows you to quickly copy all semantically meaningful settings between graph types. **Propagate To All Graph Types** will push all appropriate settings to all other graph types in one click. This can be used to quickly propagate a colour scheme or decay time preference.
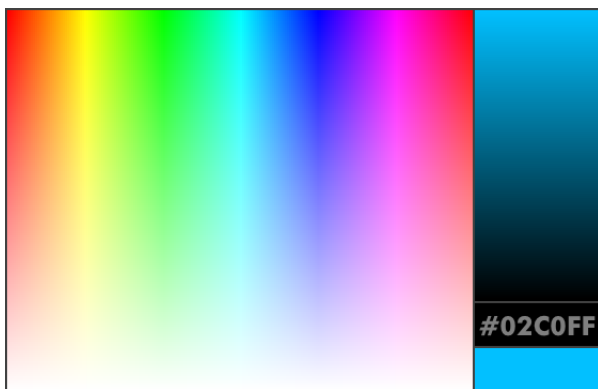
Finally, many graphs allow you to choose coloration using a colour picker. Clicking the left-hand box allows you to select a colour. Clicking in the strip on the right allows you to brighten/darken the colours displayed in the left-hand colour box. The hex code is displayed in an editable textbox for easy recall/setting, and a small swatch of the chosen colour is shown bottom-right. Clicking anywhere outside the picker will close it.

7.2 FFT Analyser

7.2.1 Overview
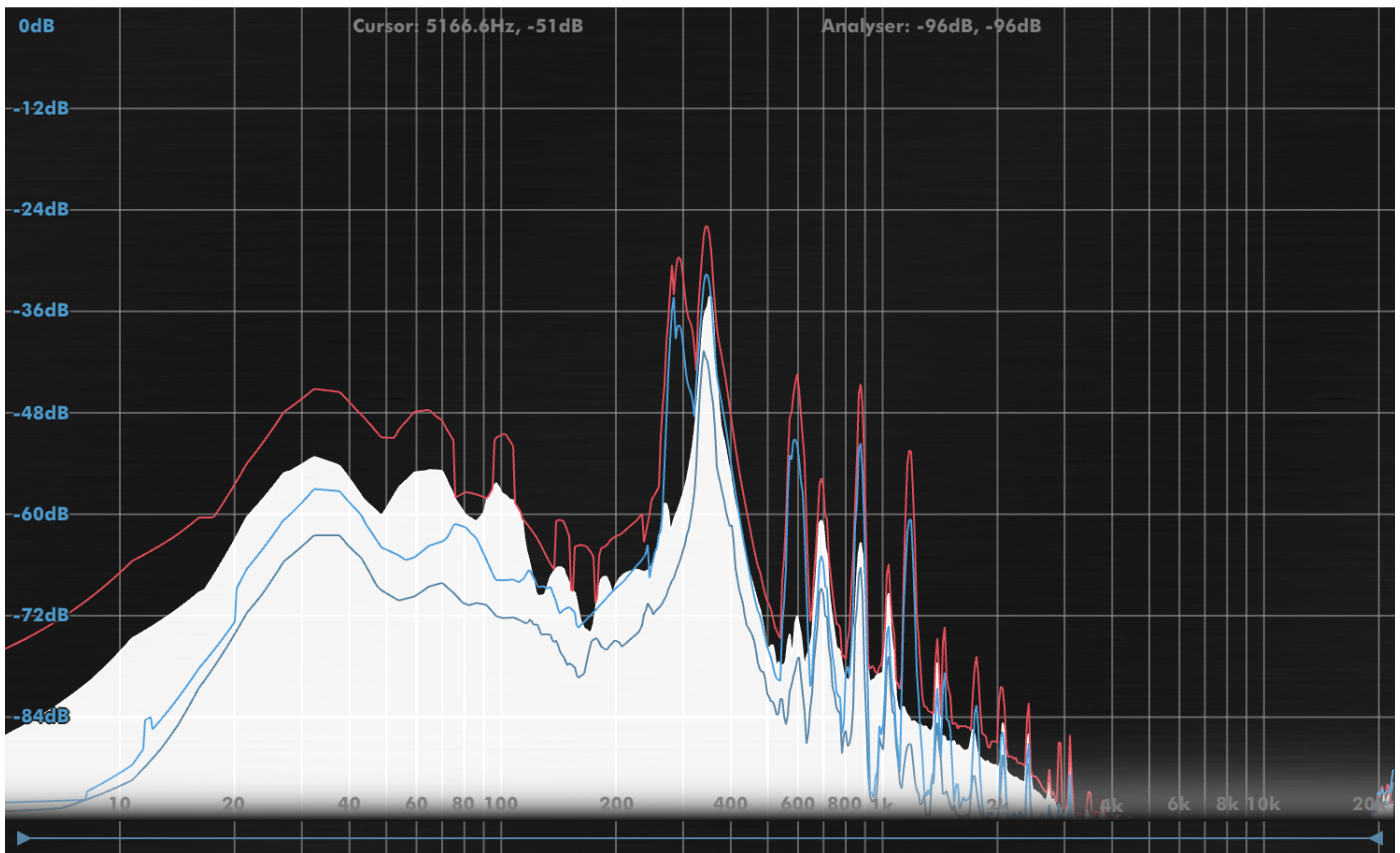
The FFT Analyser shows you the instantaneous spectral makeup of your signal. It can display two channels of information simultaneously, alongside peak information.

The analyser implemented in TrackMeter is an 8192-point overlapped-block design, using a hand-tuned Kaiser window.

7.2.2 Mouse/Wheel Control

The mousewheel can be used to zoom in and out on a region of interest.

When zoomed, dragging left-right can be used to move the range of frequencies shown.

Dragging up-down will adjust the maximum peak height shown on the graph.

The scrollbar at the base of the screen allows fine control over the zoom in frequency.

7.2.3   Preferences

These preferences form the default conventions for other graph types, so this section is worth a careful read to shed light on other types.

The rendering of the primary curve in the analyser is controlled by the (colour) Scheme preference. At its simplest, you can pick High, and use the "High Colour" colour picker to select a display colour. Lo->Hi will fade between Low and High colours based on the level of the band. By Freq will impose a hue-by-frequency scheme over the display. F&L->H (Frequency and Low->High) allows you to pick high and low colours. The hue-by-frequency scheme will be used, but the intensity of the colour will blend between the brightnesses of the Low and High colours.

Opacity sets the opacity of the filled area under the curve. Background configures the brightness of the axes behind the graph. Fade allows you to select whether the bottom of the graph is faded to enhance legibility of the frequency labelling. When disabled the labels will be stroked with a black outline.

Decay controls the rate of decay of instantaneous level. Increased decay eases readability of the curve at the expense of instantaneous accuracy. Peaks are always shown.

Max dB and Range determine the highest and lowest values shown on the graph.

Tilt will rotate the graph. Positive tilt values increase the displayed level of high frequencies. At 0dB, white noise is shown flat. At 3dB, pink noise is shown flat.

Display 1 determines whether the primary signal is drawn as outline, filled, or both.

Display 2 allows you to disable the secondary signal, or draw it as outline, filled or both. It's drawn using Colour 2.

Peaks can be hidden, shown for a number of seconds, or shown until clicked to reset.

The colours for both peak displays can be chosen with Peaks Col 1 and Peaks Col 2.

7.3 1/N Octave

7.3.1 Overview
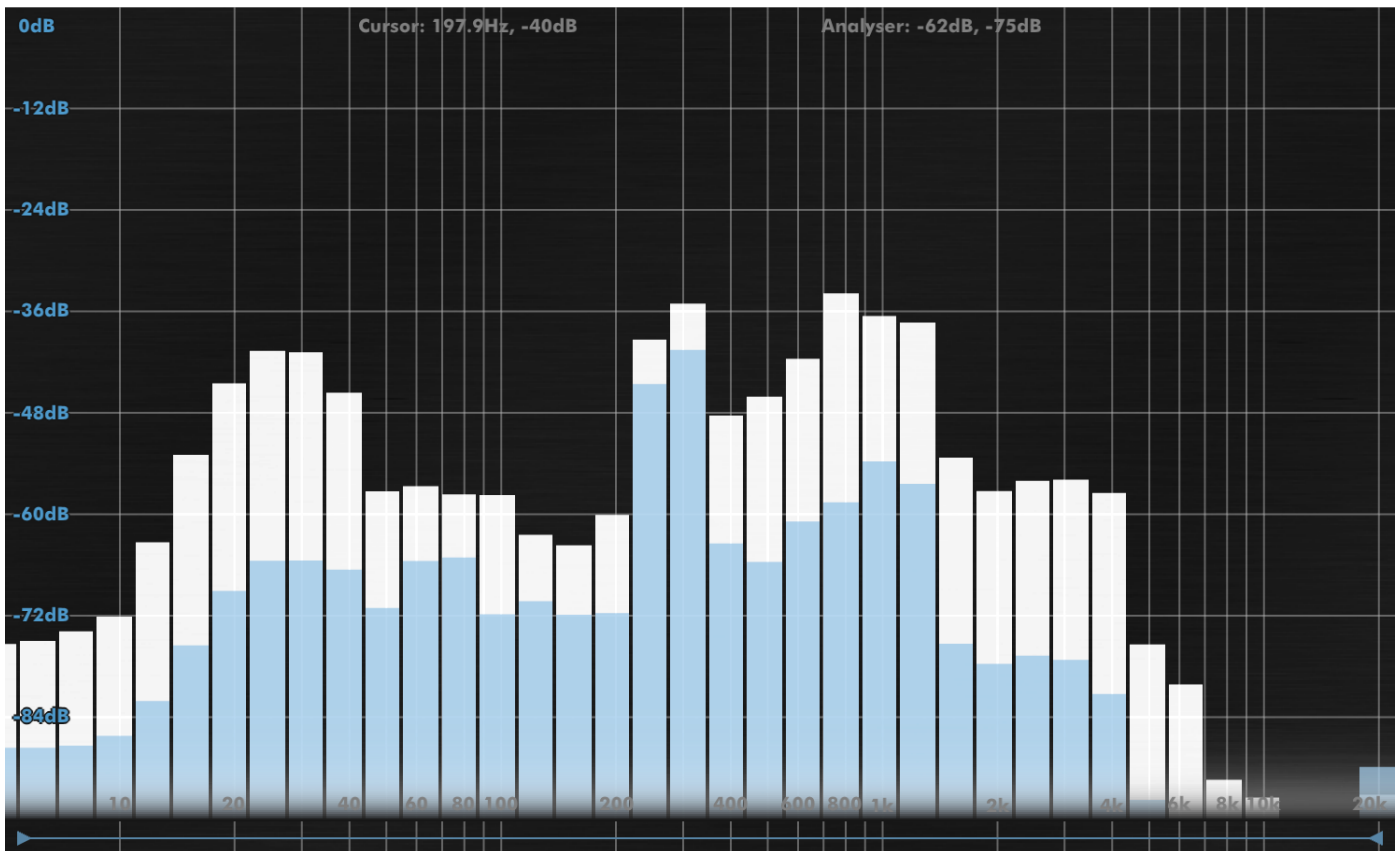
The 1/N Octave analyser is a reformatting of the data from the FFT Analyser into octave, 1/3rd octave or 1/12th octave bands. It is not an RMS analyser (described below), but is extremely convenient for quick analysis.

7.3.2 Mouse/Wheel Control

As FFT Analyser.

7.3.3 Preferences

The Subdivisions preferences allows you to select the number of divisions per octave to be shown, from 1, 3 and 12.

Alignment allows you to pick how the bands are aligned in frequency. For one octave subdivisions, the alignment determines whether the octave is considered to start at A or C, or is unaligned.

Other preferences follow the FFT Analyser.

7.4   2D Spectrograph

7.4.1   Overview
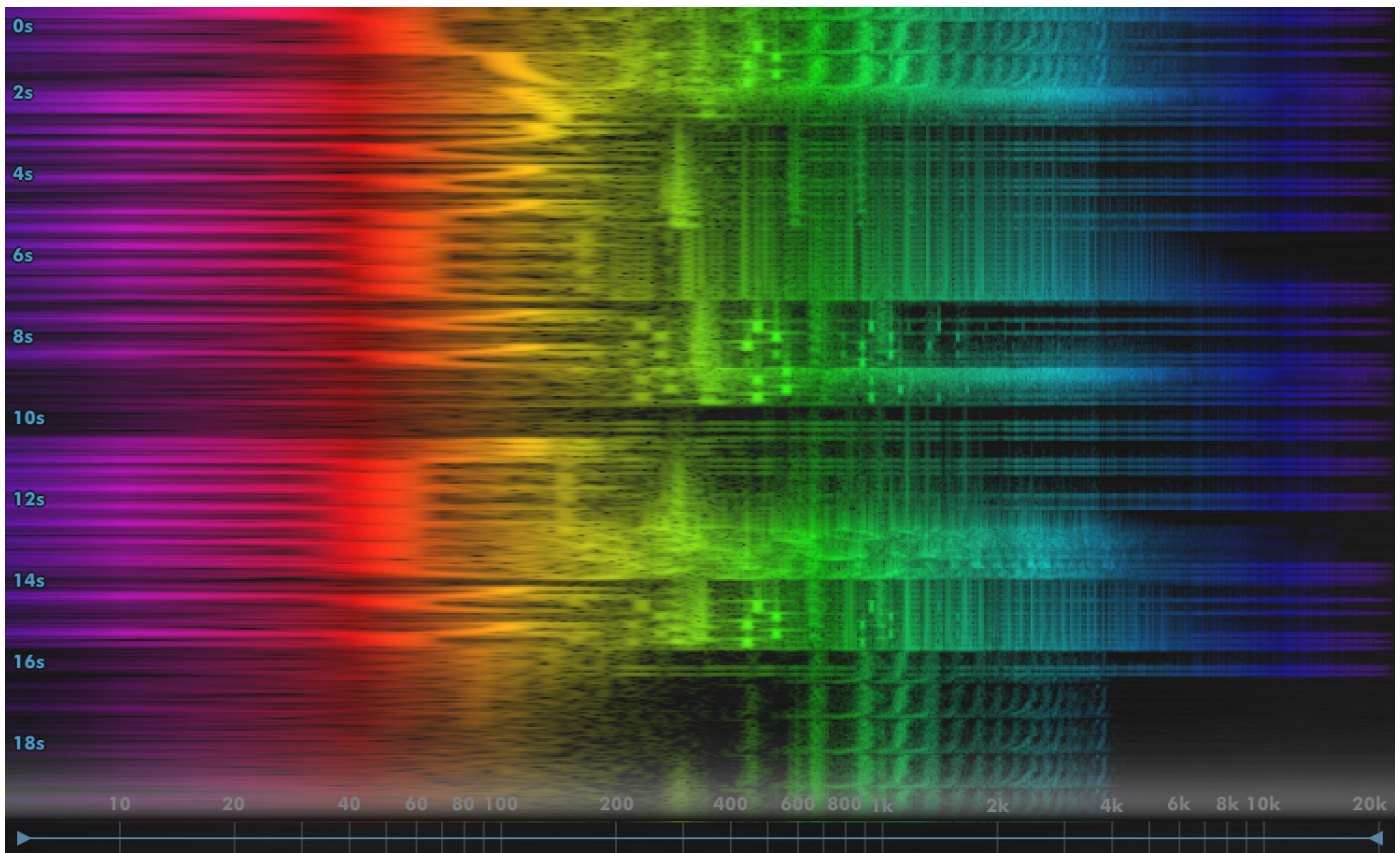
The spectrograph shows a scrolling display of frequency information over time. It can be configured to visualise individual notes, or to view broader balance of audio.

The 2D Spectrograph visualises a single mono signal.

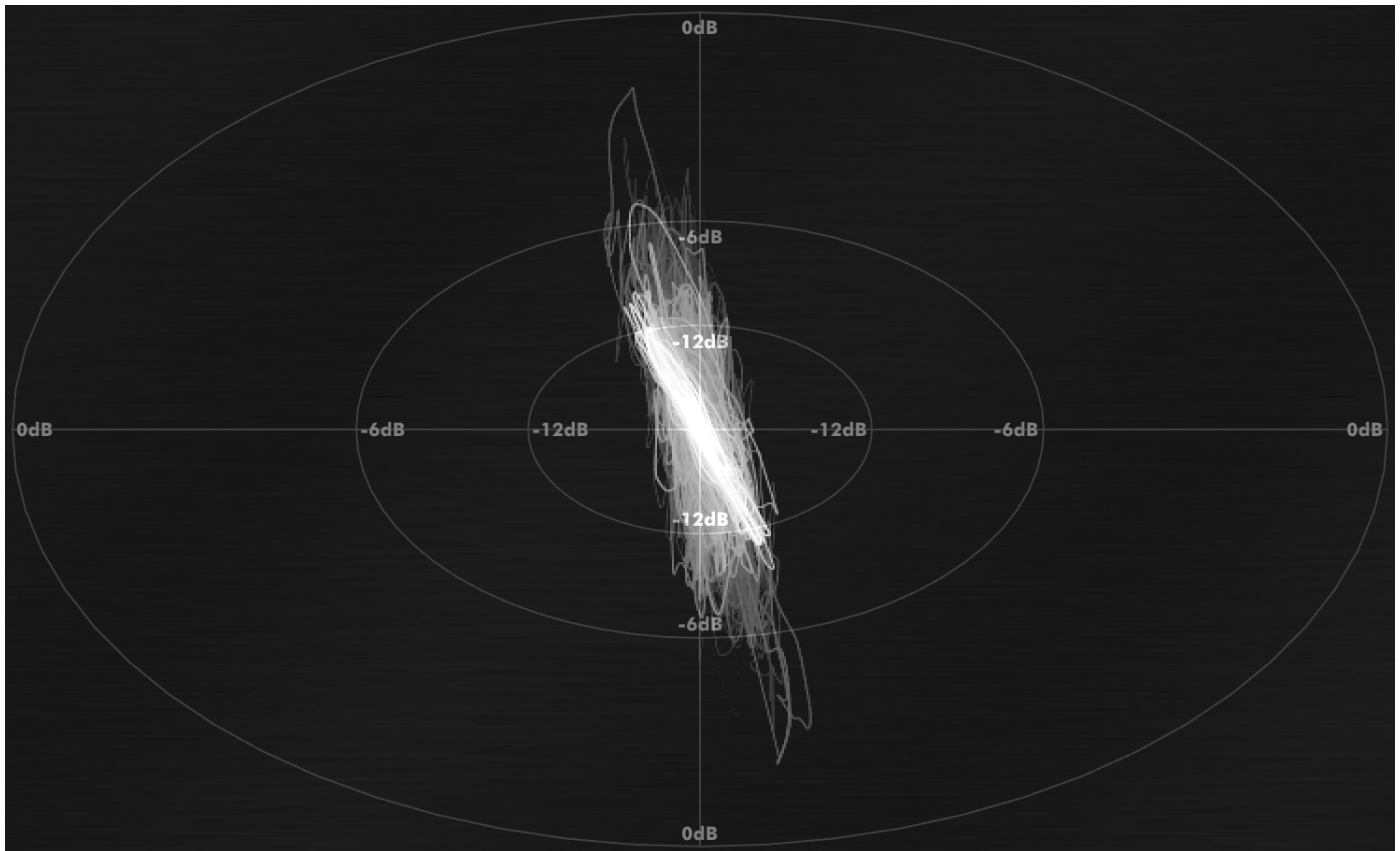### 7.4.2   Mouse/Wheel Control

As FFT Analyser.

7.4.3   Preferences

The Duration setting allows you to pick the duration of the spectrograph that's shown onscreen, in seconds.

Other preferences follow the FFT Analyser.

7.5 Lissajous

7.5.1 Overview

A Lissajous is essentially an oscilloscope where a secondary signal drives the X-co-ordinate of the oscilloscope trace. It can be used to visualise stereo width of a signal, as it forms a balance between stereo image display and a more general-purpose oscilloscope.

The Lissajous requires a stereo input signal.

7.5.2 Mouse/Wheel Control
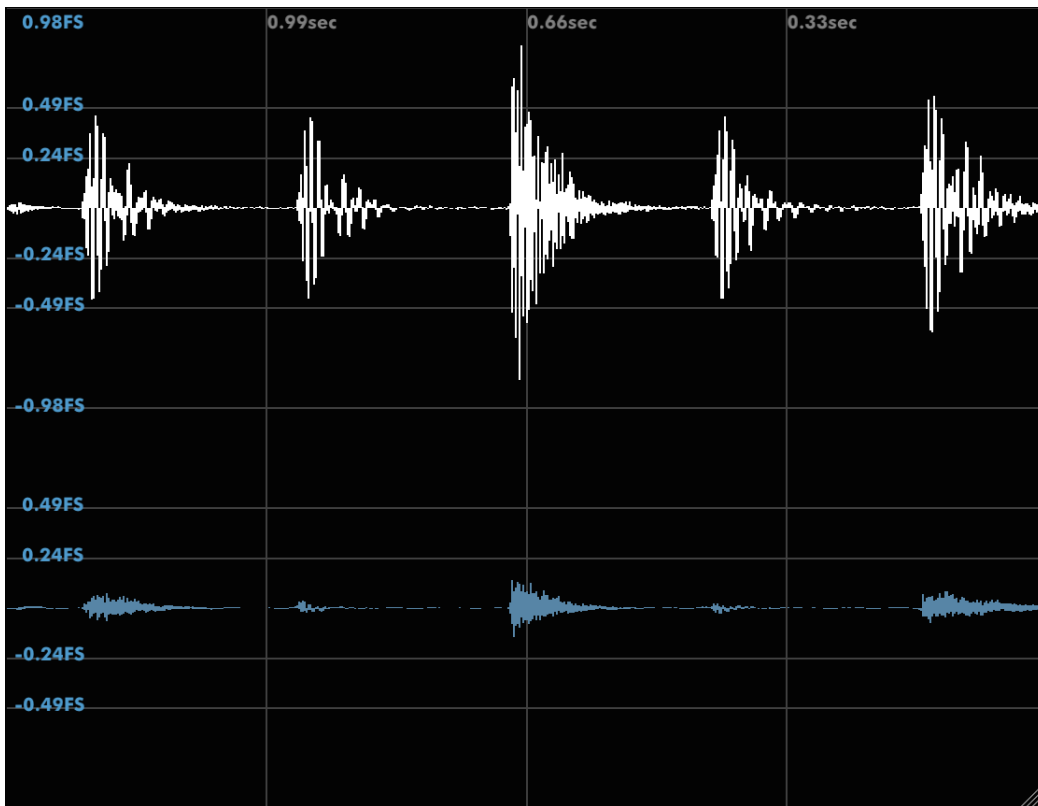
None.

7.5.3 Preferences

Decay determines how fast previous values fade from the display (analogous to phosphor fade with an oscilloscope).

The Style determines whether values are joined by lines or plotted as individual points.

Zoom allows you to adjust the display range for lower level signals.

7.6 Scope

7.6.1 Overview

The oscilloscope provides a general-purpose oscilloscope for visualisation of signals in the time-domain.

This implementation offers a variety of common uses, from triggered display of periodic signals to display of much broader durations.

7.6.2 Mouse/Wheel Control

The wheel adjusts the period of time shown on the scope.

Dragging left-right also adjusts the time period shown. Dragging up-down adjusts the vertical range of signals to be drawn. When a trigger is in use, dragging up-down allows setting the trigger level.

7.6.3 Preferences

Max dB reflects the range of level that will be shown on the scope. This value is adjusted by dragging up-down.

Trigger engages a traditional scope trigger. It is triggered exclusively by the first input signal, and can be either Rising or Falling. When one of these options is selected, the trigger level is drawn as a horizontal line on the scope.

7.7 3D Spectrograph

7.7.1 Overview
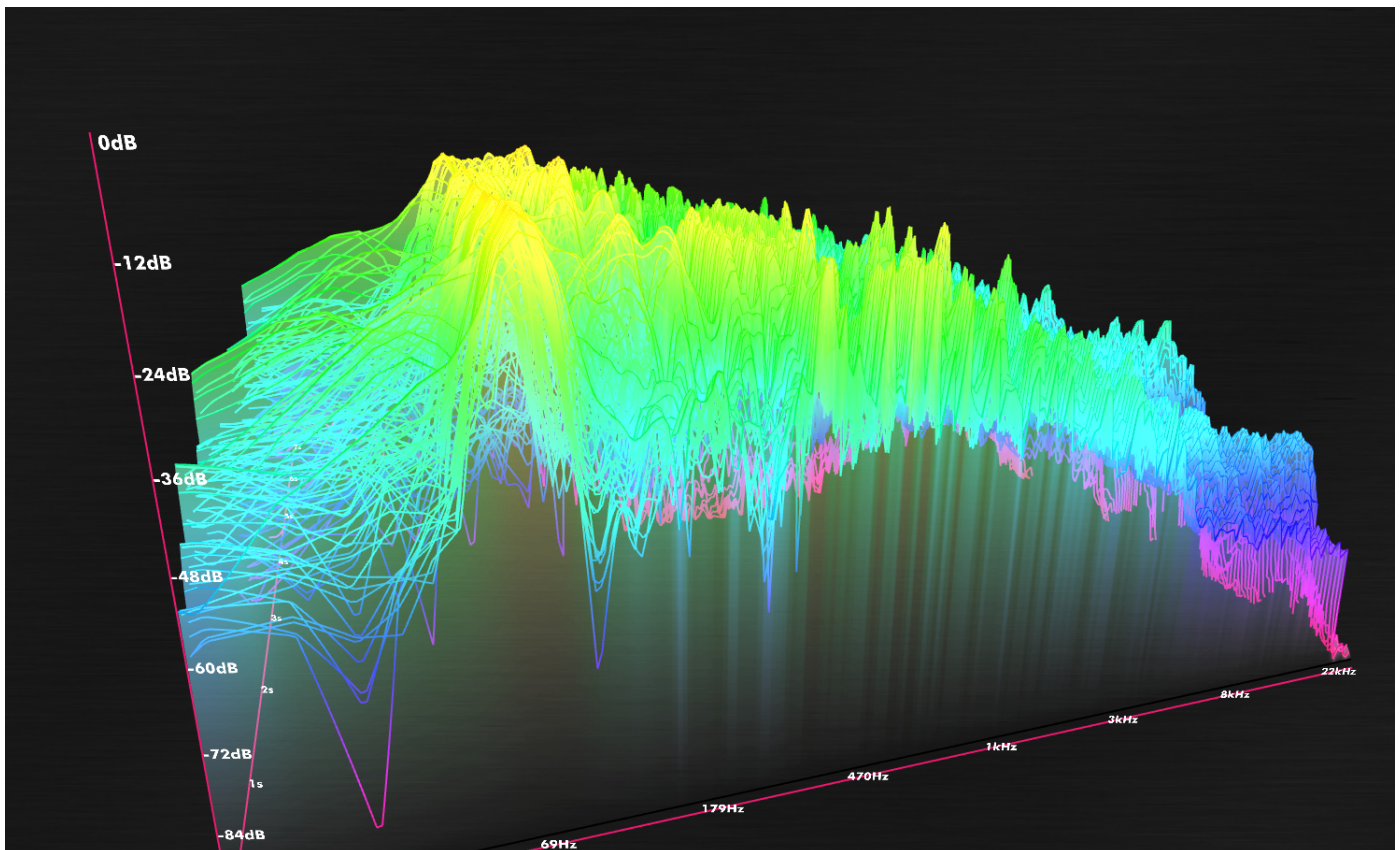
The 3D Spectrograph is a three-dimensional rendering of the 2D Spectrograph, which allows for a surprising amount of insight into the spectral and temporal makeup of a signal. It renders a single mono signal.

It is possible to lose significant amounts of time to this type of analysis, and it's likely that we would have been able to release the product many months sooner had we not been staring at this graph. Gawping.

7.7.2 Mouse/Wheel Control

Wheel zooms. Mouse up-down tilts the display up-down. Mouse left-right rotates the image.

7.7.3   Preferences

Colour scheme and analysis preferences follow the FFT Analyser conventions.

The shape rendered is composed of Surface and Mesh. The surface can be disabled to render only the Mesh, can use Blackout style to provide opacity for the Mesh, or can be rendered as a full Surface. The Mesh can be disabled, rendered as ridges along the frequency axis, or displayed as a wireframe net. Both Surface and Mesh have Opacity settings.

Axes determines whether the axes are shown or hidden.

Z-Scale allows stretching of the time-axis onscreen to fit the aspect ratio of your display, or to allow you to zoom deeper in.

Time scales the duration of audio to be shown onscreen.

Freq Sub allows you to reduce the resolution of the frequency information for simpler readouts at the expense of accuracy.

7.8   Phase Meter

7.8.1   Overview
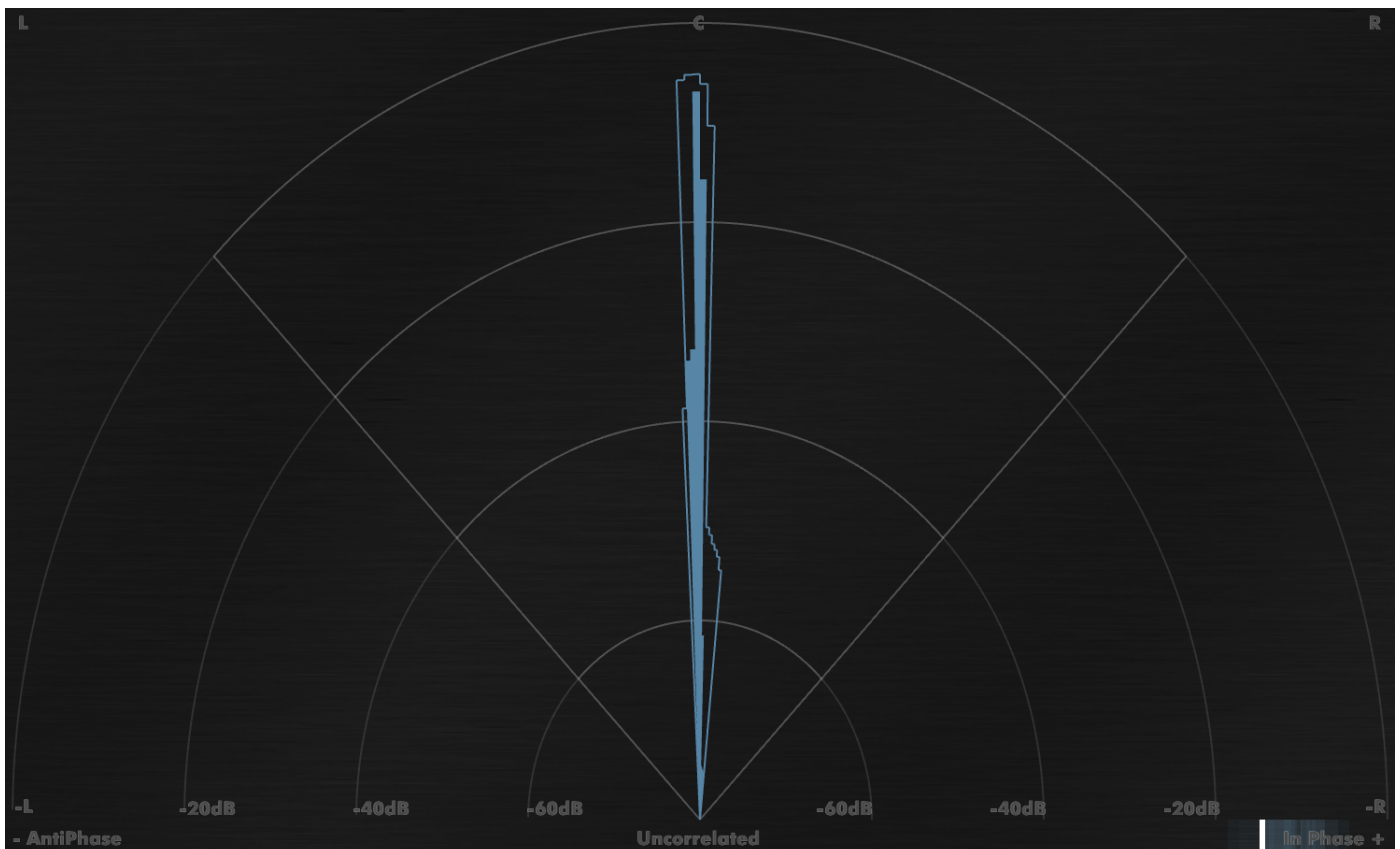
The Phase Meter provides a map of the stereo field, so you can watch stereo movement of elements in a signal. The display renders directly up as mono, hard left and hard right at 45 degree offsets, and anti-phase data to the extremes. It requires a stereo signal source.

At the base of the Phase Meter is a small correlation meter to show the correlation of the signal.

7.8.2   Mouse/Wheel Control

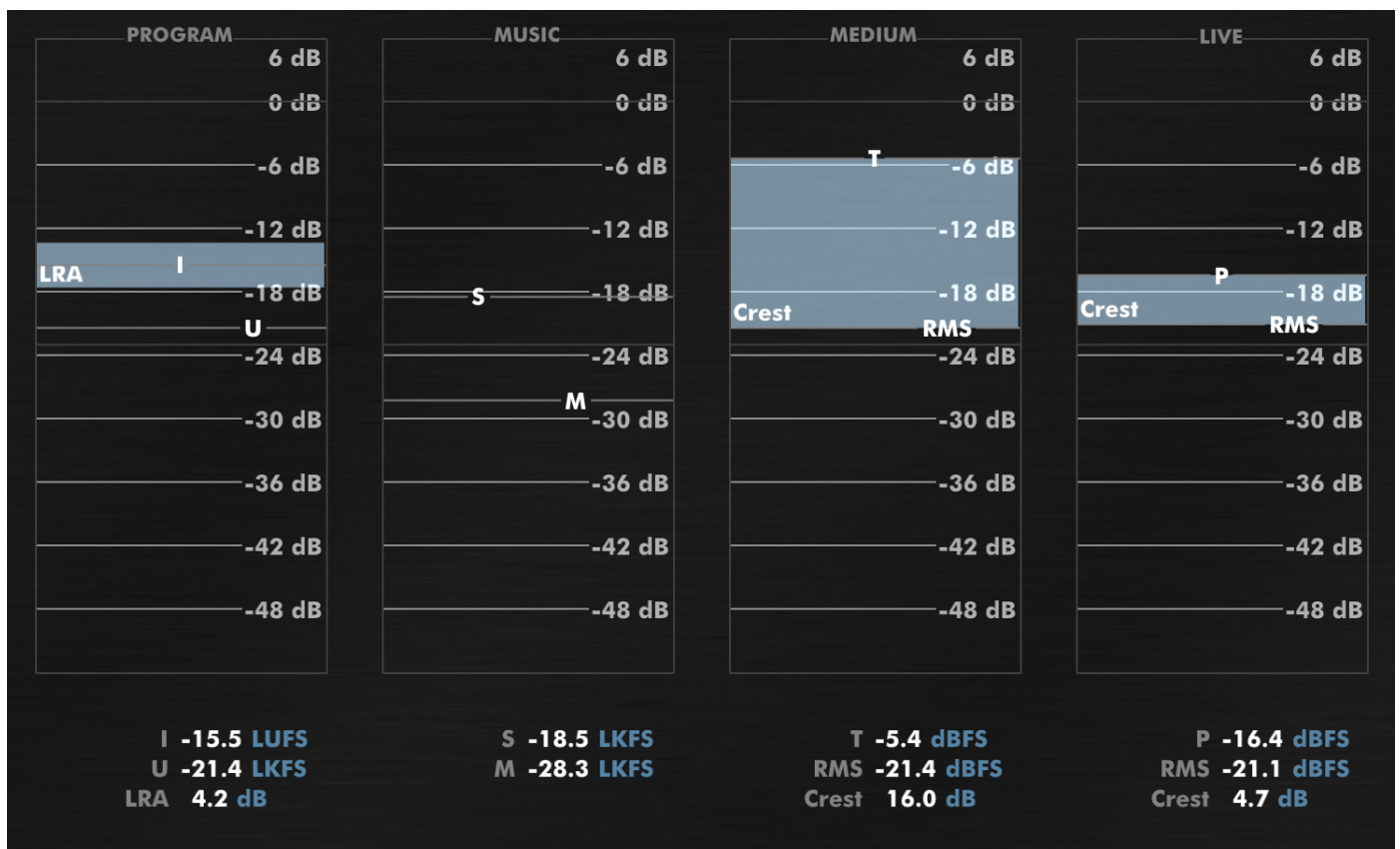Clicking resets any peak values shown.

7.8.3   Preferences

The Decay setting determines how quickly old values decay from the meter.

Peaks can be shown, and Peaks Colour determines their colour.

7.9   Loudness Meters

7.9.1   Overview

The Loudness Meters provide a set of statistics for your audio, from conventional Peaks to the new ITU1770 loudness standard figures. It is split into four domains.

Program provides the EBU r128 Integrated loudness statistic (I), the ATSC A/85 Ungated loudness statistic (U), and the EBU r128 LRA statistic.

Music provides the ITU1770 Short-term (S) statistic, and Momentary (M) statistic.

Medium provides the ITU1770 True Peak (T) statistic, includes the ATSC A/85 Ungated loudness statistic as RMS, and displays the Crest factor as their difference.

Live provides a traditional PPM and RMS measurement, and displays the Crest factor as their difference.

7.9.2   On Loudness

In case these specifications don't mean anything to you yet, we provide here a very quick introduction to these terms. Traditionally, audio metering has been based on PPM or RMS meters (also provided here), which respond to voltage, and do not provide a good perceptual match to human experience of loudness. The ITU set about working out a remedy to this. The first stage towards this was to define more precisely how to take an RMS measurement, and to compensate for the fact that different frequencies are experienced as having different loudness.

The latter part was resolved with what's known as the K-weighting filter, which includes a high-pass filter at roughly 60Hz, and a high shelf at about 1.5kHz with about 4dB of boost. This simple filter has proven remarkably effective in listening tests at compensating for the frequency-nonlinearity of the ear.

The RMS measurement was now to be performed on K-weighted audio. Signals were to be processed per-channel, and then squared and summed. The RMS mean is computed over some number of samples. When that number of samples equates to 0.4 seconds, we call that the Momentary (M) statistic. It has the feel of a traditional PPM meter, but maps better to perceived loudness. When that number of samples corresponds to 3 seconds, we call that the Short-Term (S) statistic. It has the feel of a traditional RMS meter. The benefits are increased correlation with human experience of loudness, and unambiguous, clear definition as to how they are computed.

You can watch and compare the Music and Live sections to compare the new ITU1770 measurements with traditional Peak/RMS readings.

The ITU (and the EBU and the ATSC and many others!) also had a requirement to be able to measure the loudness of an entire program with a single number. Historically this would literally have been a TV program or radio program. It has since expanded to film, and it is expected to transition to music very soon.

Obviously a peak or an RMS meter respond in real-time to audio information, so some further processing was required. The ITU came up with two solutions. One was to compute the RMS of the entire K-weighted (filtered) signal. This is the Ungated (U) signal, as used by ATSC A/85, and is written down as a figure in LKFS (with the understanding that LKFS means dBFS for a signal that has been measured as ITU1770 ungated).

The second solution the ITU proposed was to record all Momentary (M) readings for the duration of the program, at a minimum of ten per second. Any readings below -70dB are discarded, and the average of what remains is computed. Then subtract 10 from that figure, and discard any readings below that number, and recompute the average. This is known as the Integrated (I) (gated) loudness, as used by EBU r128. It is written down as a figure in LUFS (with the understanding that LUFS means dBFS for a signal that has been measured as ITU1770 integrated).

Whilst I and U figures are displayed in realtime, they are only ever meaningful once an entire program has been played through them.

The EBU wanted a further statistical measurement for a complete program, to measure the perceived dynamic range. This is the LRA statistic. It is measured by recording all Short-Term (S) readings for the duration of a program and discarding any readings below -70dB. They are then compiled into a histogram, where the top 5% and bottom 10% are discarded. The difference between the highest and lowest values remaining is the LRA figure, and is a figure in dB (no units, since it's a ratio).

Finally, True Peak is an estimation of the maximum instantaneous level of a digital signal after it has been played through a DAC. It is possible for a digital audio signal which does not clip to represent a signal that does clip in the analogue domain. This is measured by simply upsampling the audio and measuring peak levels.

The TrackMeter loudness meters pass all ITU1770 and EBU r128 calibration tests.

7.9.3   Mouse/Wheel Control

None.

7.9.4   Preferences

Fill Colour sets the colour for statistics drawn as boxes (LRA/Crest).

Label Colour sets the colour for the readout text.

Target allows you to set the Loudness Target calliper for the graphs.

Clip Level allows you to set the Clip Level calliper for the graphs.

Marker Opacity sets the opacity for the markers behind the meters.

Meter Range and Meter Top allow you to configure the meter gain range.

Meter Size allows you to switch the view so to smaller meters and larger text beneath.

7.10 Loudness History

7.10.1 Overview

The Loudness History allows you to visualise the ITU1770 K-weighted Short Term loudness over the duration of a block of audio, combined with an EBU r128 histogram of the LRA data. These terms are discussed in the Loudness Meters section. All values shown are LUFS ITU1770 K-Weighted Short-Term measurements.

Loudness History measurements never translate to accurate ATSC A/85 or EBU r128 integrated statistics. They provide a way to measure perceptually-weighted loudness over the course of a recording, for the purpose of providing feedback that can be used to dynamically process data towards meeting your targets.

The Loudness History requires a stereo input signal.

7.10.2   Mouse/Wheel Control

Dragging left-right adjusts the time-scale shown. Dragging up-down allows you to move the target loudness calliper.

7.10.3   Preferences

Colour 1 sets the colour for the Short-term loudness curve. Colour 2 sets the colour for the Variance window. Opacity configures overall drawing opacity.
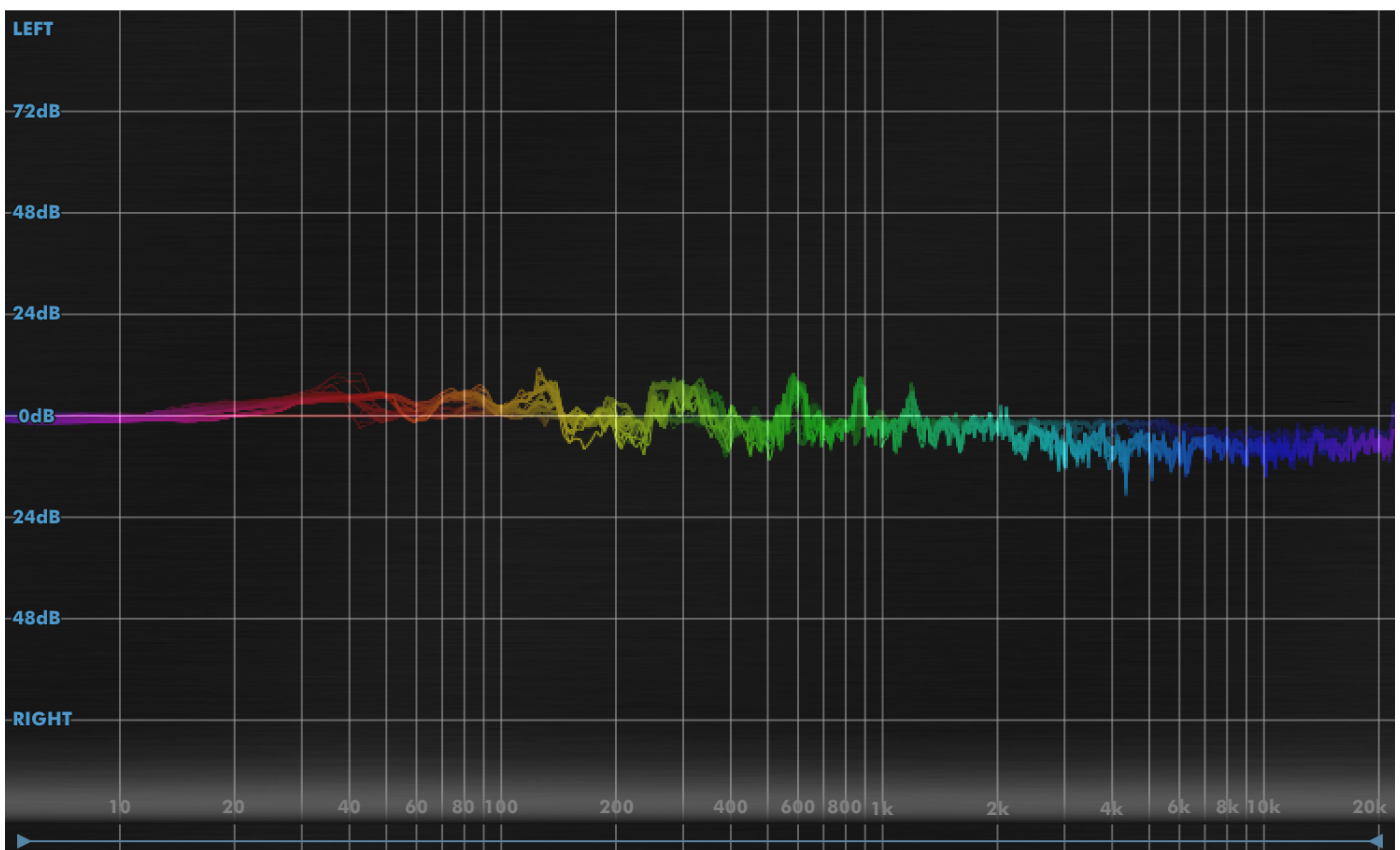
Target sets the loudness target calliper level, which is displayed over the graph for easy checking of when the signal is above and below target.

Labels set to Abs labels the graph relative to 0dB. Set to Target, the labels are relative to the target loudness calliper.

Max dB and Range set the range of loudness shown.

7.11   Phase Spectrum

7.11.1   Overview

The Phase Spectrum renders the instantaneous difference in stereo position by frequency.

It can be used to see signals jump left or right, and to see their spectral shape as they do. It also provides a very clear display of stereo width by frequency.

The top of the graph represents 'leftness' and the bottom 'rightness'.

7.11.2   Mouse/Wheel Control

As FFT Analyser, without up-down drag.

7.11.3   Preferences

The Mode picks whether you want to see the offset in dB, or as a correlation percentage.

Like the Lissajous, the drawing can be joined lines or dots. The Fade Out parameter also determines the fade time for old drawn values.

In dB mode, the Distance displays the dB distance scale between Left and Right, allowing you to go from broad scale to zoomed right in.

Other preferences follow the FFT Analyser conventions.

7.12 RMS Analyser

7.12.1 Overview
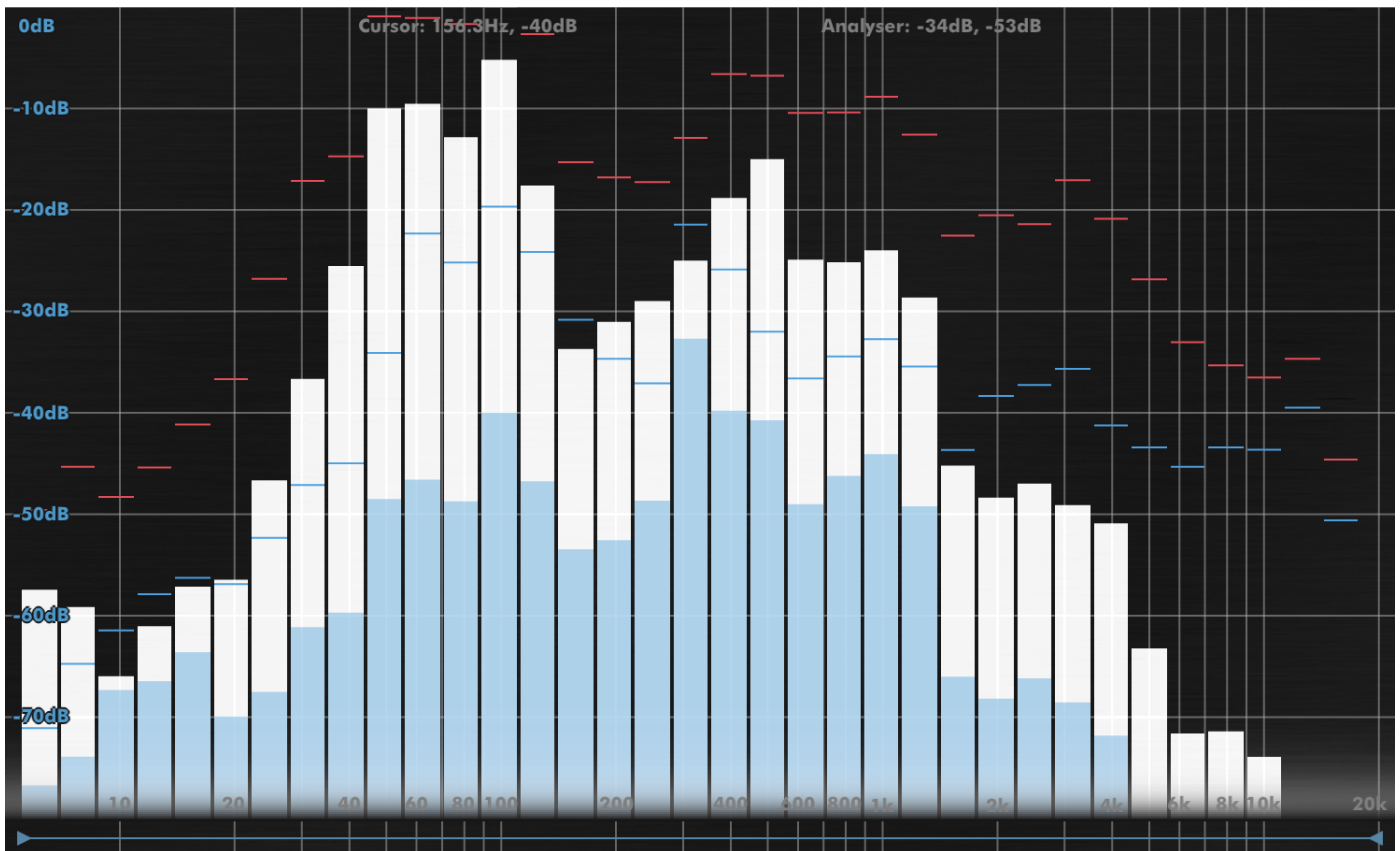
The RMS Analyser is a classic 1/3rd octave analyser design based on overlapping 36dB/oct Butterworth Band-pass filters, with per-band peak detection. This has the advantage of even resolution across the spectrum, and also allows for measurement of per-band crest factor.

The analyser has been normalised to display white noise as flat. This can be reconfigured using the Tilt preference.

7.12.2 Mouse/Wheel Control

As FFT Analyser.

7.12.3 Preferences

Align allows you to choose the alignment of the filterbank. Filters can be centered to align with 1kHz, A (440Hz) or C (261Hz).

Timebase determines the timebase used for the RMS computation. Low values provide very responsive measurements but are harder to read.

Display and Analysis preferences follow FFT Analyser conventions.

The Mode preference determines whether the graph displays conventional analyser curves, or a Density display, which shows the distance from peak to RMS reading (akin to crest factor) per-band. This offers a measurement of the density of a frequency band over the duration of the RMS timebase, which provides valuable information for mastering work.

7.13   Tuner

7.13.1   Overview
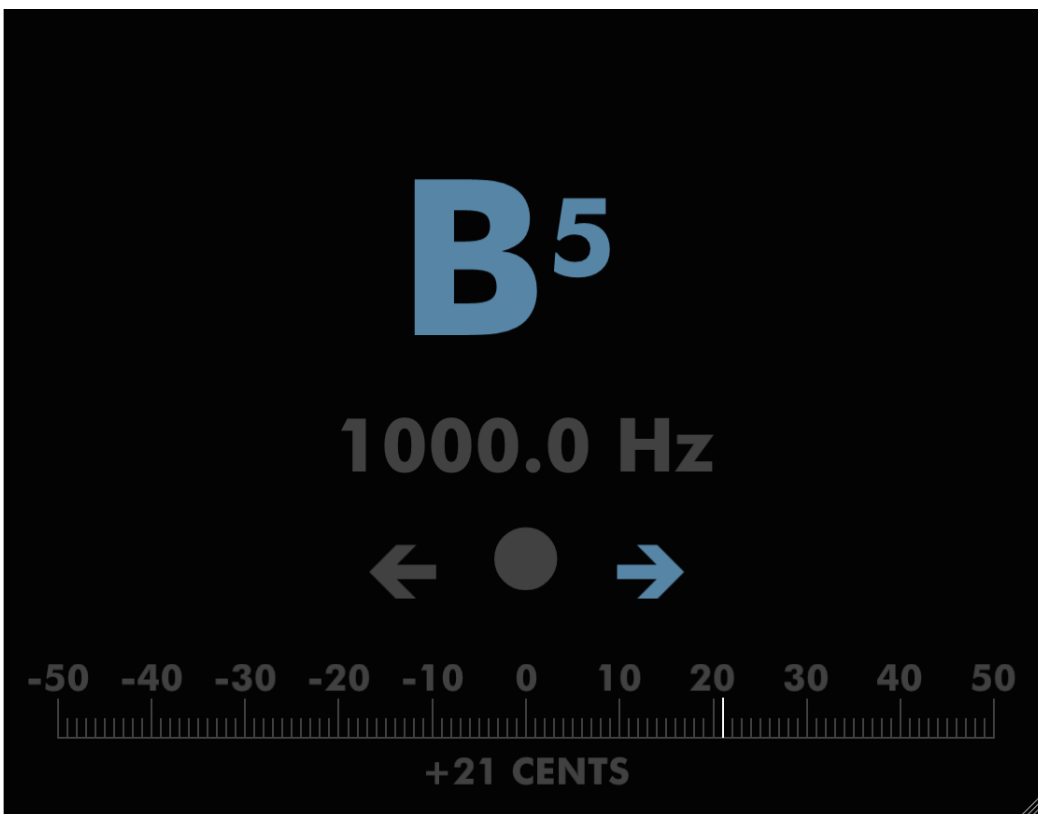
The Tuner is a high-precision 440Hz Equal-Temperament Chromatic tuner.

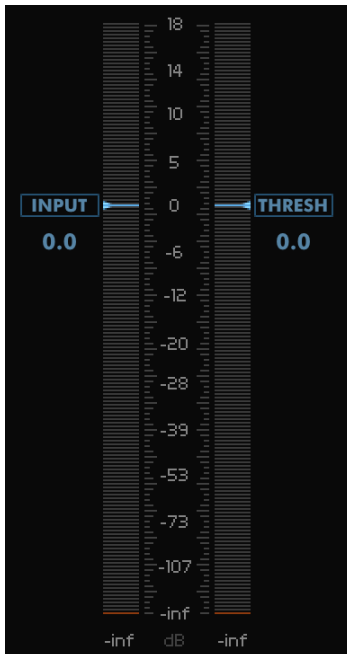### 7.13.2   Mouse/Wheel Control

None

7.13.3   Preferences

Display colours may be adjusted.

8   Main Controls

8.1   Controls

Callipers

Callipers are calibrated vertical sliders which allow you to precisely set values for a plugin. They are positioned over a PPM meter to allow you to set a level relative to the signal which it relates to.

The current value is displayed in a text field to one side of the indicator line. This can be double-clicked or dragged to alter the value.

You can shift-drag to fine-adjust a calliper, adjust using the mousewheel, or double-click to have it jump to a specific value.
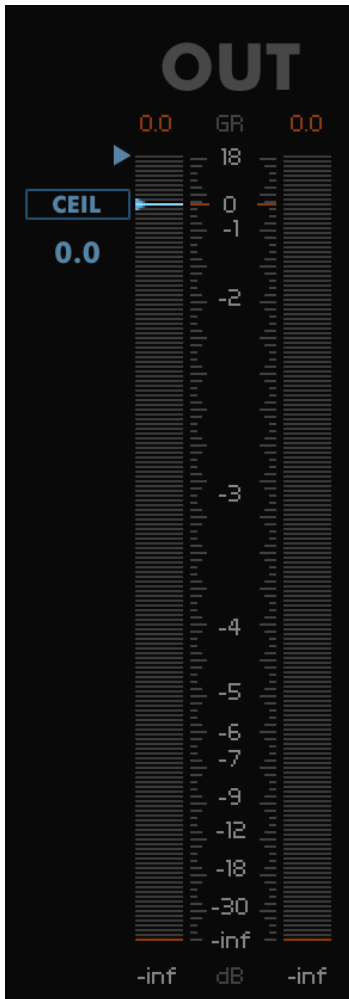
Some Callipers have a small triangle at the top-left. This indicates that they can be zoomed for precision work. Clicking the triangle alters the scale to zoom in on the region of interest. Functionality continues to work exactly as before, albeit allowing more precision over the range shown.

Sliders

Sliders are used for adjusting the most important settings.
They can be dragged to adjust, or shift-dragged for fine-adjust.
Double-clicking will jump the slider position to wherever you click.
Sliders can be adjusted using the mousewheel.

Text Values

Text values are rendered in blue, and followed by a unit in darker blue.
They can be adjusted by dragging with the mouse or shift-dragged for fine-adjust.
When double-clicked, you will be able to enter text. Press enter to confirm the new value.
Text values can be adjusted using the mousewheel.

Dropdown Menus

Clicking a Dropdown Menu allows you to select from a discrete list of options.
You can cycle through options using the mousewheel.

MIDI Learn
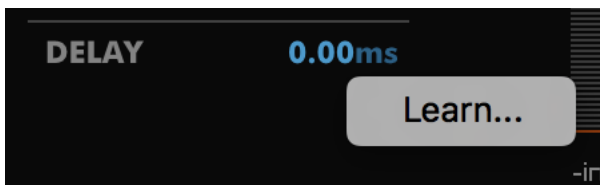
You can right-click (or Ctrl+Click on mac) any control to open the MIDI Learn menu. Selecting "Learn..." will wait for MIDI CC input and bind the control to the first MIDI CC controller received. Parameters are displayed in green to indicate that they have been MIDI Learned. MIDI Learn mappings affect all instances of a plugin and are saved globally, making it easy to map the plugins to your MIDI Controller. Right/Ctrl-clicking a Learned parameter displays a menu with information about the mapping and an "Unlearn" option to forget the MIDI mapping.
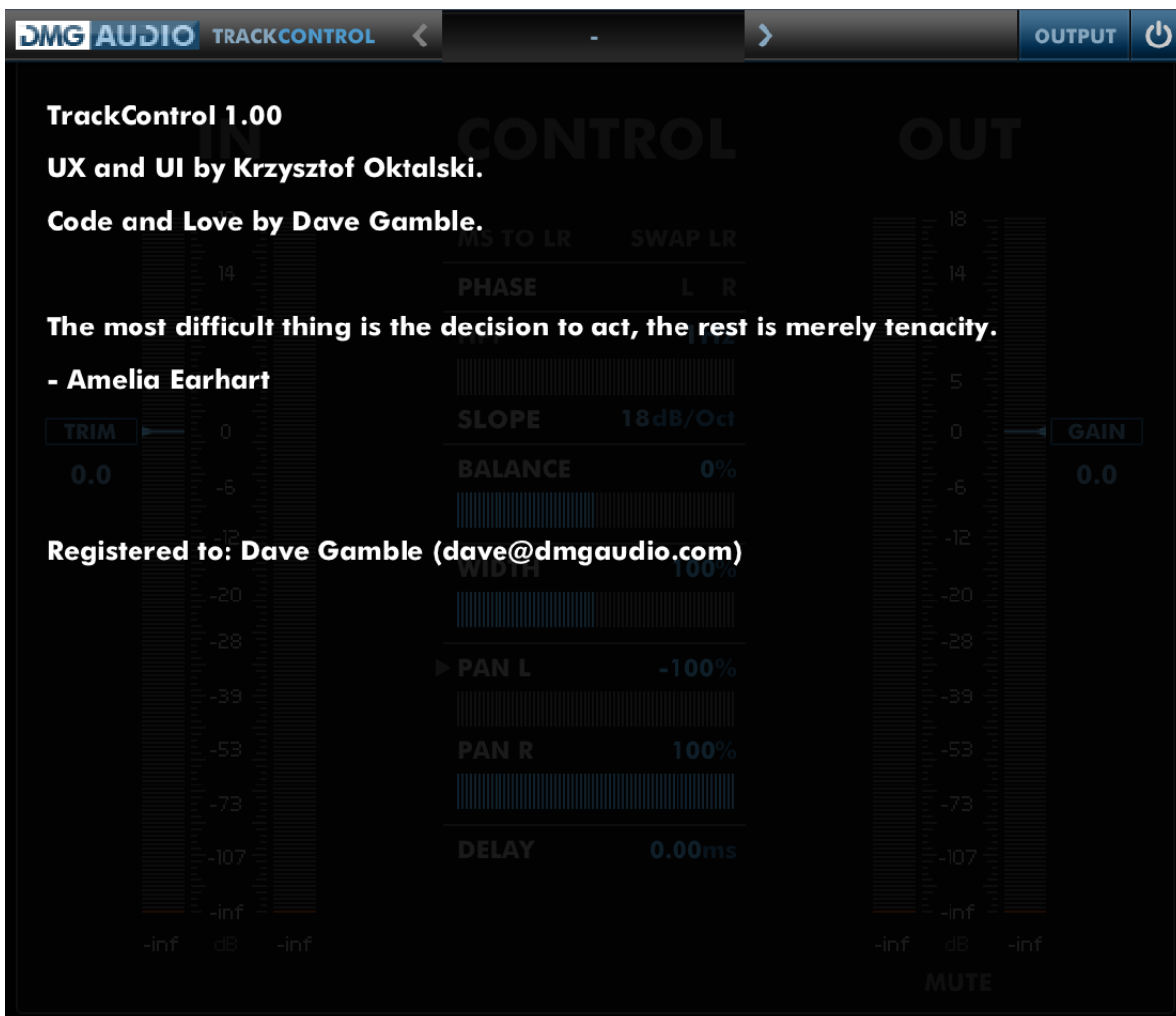
9 Other UI

9.1 The Menu Bar
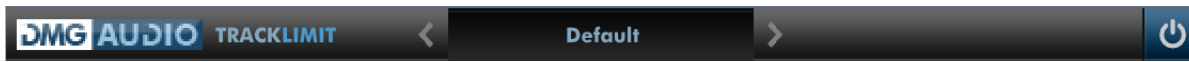
Each plugin has a menu bar at the top of the plugin UI, which may contain plugin-specific controls described in that plugin's section. In this section we describe the common menu bar controls.

9.1.1 DMGAudio Logo

Clicking the DMGAudio logo opens up the "About" page below:

On this page you can see the current version number, credits, and your license registration information. If you don't have a license installed, or your demo has expired, that information will be displayed here.

9.1.2   Bypass

The Bypass button bypasses all processing within the plugin and routes the input directly to the output.

9.2   Presets
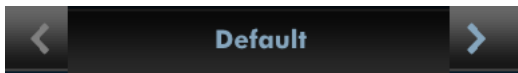
Each Track Plugin includes a set of presets, which you can add to, to build up your library of settings.

By default, on OS X they reside at /Library/Application Support/DMGAudio/Track <plugin>/Presets/ and on Windows, at C:\Users\<you>\Documents and Settings\DMGAudio\Track <plugin>\Presets\.

You can change the location of the presets on Windows by re-running the installer and changing the "DMGAudio Track <plugin> resources folder" path during the install.

Saving presets into these folders will allow you to select them from the Preset menu.

Don't forget to include these files in your backups.

9.2.1   Previous / Next

These buttons allow you to quickly jump back and forward within the list of presets.
They grey out to show you when there are no further presets available for selection.

9.2.2   Preset Menu / Name

This shows the Preset name which can be modified with a right-click. Clicking here will open the Preset Menu, which is a multi-level menu of all of the presets you currently have saved.
You can arrange this folder to suit your needs, but be aware that installing updates will reinstall the factory set.
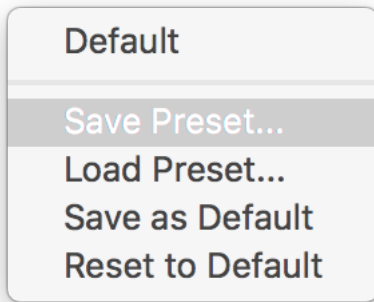
Selecting "Save Preset..." will allow you to save your current settings as a preset. By default this will save into the presets folder, so that the preset will appear on the menu.

Selecting "Load Preset..." will open a file browser to allow you to load a preset from anywhere on your system.

Selecting "Save as Default" will store your current settings as the default settings for new instances, or to be used when selecting "Load as Default".

Selecting "Load as Default" will restore the settings of this instance to the last saved default settings, or to factory settings if you haven't saved your own default.

10   Troubleshooting

10.1   Get in touch

If you run into trouble, get in touch! We're here to help, after all!

For technical support, feature requests and general discussion, visit:
https://dmgaudio.com/support.php

Or you can just email us directly at: contact@dmgaudio.com

For support with DMGAudio software, sending email is the way to go. It allows us to be much faster and more responsive, and we try to spend more time answering our emails than hanging out on social media!

For general discussion though, please feel free to drop by on Facebook/Twitter and tell us how you're getting on. We'll be posting news about updates there too - but also do join the DMGAudio mailing list to make sure you never miss out.

 https://facebook.com/dmgaudio

 https://twitter.com/dmgaudio

 https://www.youtube.com/c/dmgaudioofficial

Questions about licensing, license transfers and discounts are covered in our FAQ.

Information about version numbers, updates, features and bug fixes is listed under the Specifications tab on the Track Range Product Pages.

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