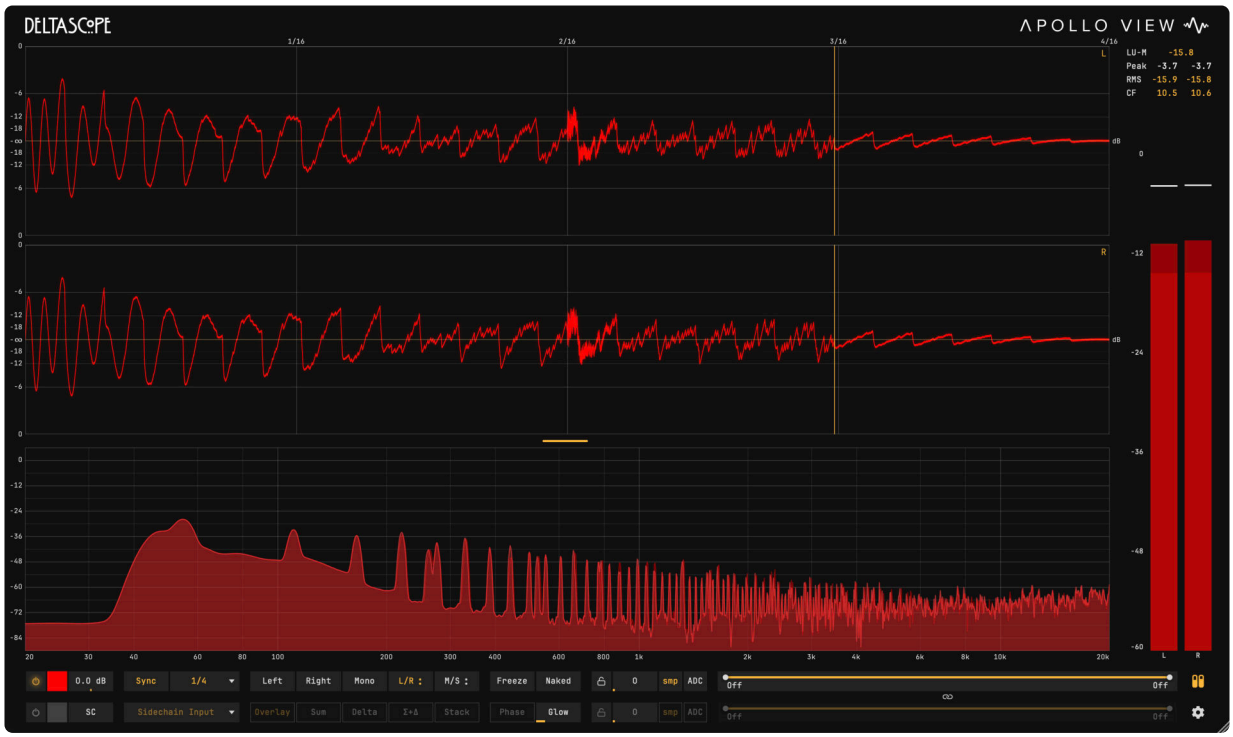
DELTASCOPE

SEE THE DIFFERENCE



MANUAL V1.4

APOLLO VIEW

TABLE OF CONTENTS

INTRODUCTION 3
What is it? 3
Why is it useful? 3
What it does to your audio 3
FEATURES 5
System requirements 6
Ableton Live on Windows 6
QUICK START 7
Your first comparison 7
OVERVIEW 8
Key 8
Resizing 9
Slider gestures 9
Settings at a glance 9
TIMEBASE 10
Sync 10
Free-run 11
MIDI 12
THE DISPLAY 15
Two panes, one divider 15
Rolling mode 16
Clip indicators 17
POWER, GAIN AND COLOUR 18
Power 18
Gain 18
Colour 18
Colour by frequency 20
CHANNEL MODES 24
One lane or two 25
FREEZE AND NAKED MODE 26
Freeze 26
Naked mode 26
ZOOM, PAN AND MEASUREMENT 28
Zoom and pan 28
Measurement 28
SPECTRUM ANALYSER 30
Ranges and the Nyquist line 30
Resolution and Multi FFT 31
Colour by frequency in the spectrum . . 33
Crosshair measurement 35
Distortion readout 36
Peak hold 41
FFT settings (Settings > FFT) 42
SIDECHAIN 44
Choosing the source 44
Comparison modes 46
Glow 48
DELTA AND HEAR THE DELTA 49
Reading a delta 49
The perfect null 49
Per-lane delta 51
The delta in the spectrum: Difference and Magnitude 51
Tracing a processor's real curve 52
Hear the Delta 53
Hear the Sum 55
ALIGNMENT: OFFSET AND ADC 56
Latency between two linked instances . 57
ADC on one instance 57
Offsets on a linked pair 58
ADC on a linked pair 60
Finishing by eye 62
PHASE CORRELATION 63
Phase settings (Settings > Phase) 64
BAND FILTERS 66
LEVEL METERS 68
Meter settings (Settings > Meters) 69
USE CASES 70
Fix kick and bass phase 70
See what your processing actually does 72
Listen to the difference 72
Check what survives mono 72
Deliver to spec 73
Make video content 74
SETTINGS 75
What is remembered 76
Files 77
DEMO MODE 78
LICENSING 79
IN SHORT 80
LICENCE AND WARRANTY 81

INTRODUCTION

WHAT IS IT?

Oscilloscope and spectrum analyser, with a second input and a difference view.

DeltaScope draws your audio as it happens. The waveform is at the top of the window and the spectrum analyser is below it. A new instance starts slim, showing the scope alone; the analyser appears as soon as you make the window taller. The stretch of audio on screen can follow the beat, span a set length of time, or lock to the cycle of the note you're playing.

DeltaScope can also show a second signal at the same time: a reference from the sidechain input, or from another DeltaScope anywhere in the project. You can overlay the two, stack them, add them together, or show the delta: the difference between them.

Most of the time you'll live in the scope and the analyser. The delta is there for the moment you need to see what a processor is doing.

WHY IS IT USEFUL?

A scope shows what a meter can't: the shape of a sound, whether a hit lands on the grid line or after it, and whether the low end is one clean wave or two waves cancelling each other. It also shows clipping as it happens: the display edge lights red when samples rise above 0 dBFS, and a signal that something upstream clipped flat shows its flat tops. The analyser shows the same audio by frequency. The delta shows what a plugin is actually changing. Subtract the original from the processed signal and only the change is left:

- **An EQ**: the delta shows what the curve is really doing, in level and in timing.
- **A compressor**: the delta is what the compressor is changing, drawn in time: its attack and release, as a waveform.
- **Which frequencies change most**: the spectrum of the delta shows where the change carries the most energy; the Magnitude view traces the processing's actual frequency response, with no test signal.

DeltaScope is made by Apollo View Modular. We build Eurorack modules and audio plugins, and we like a tool that knows its job and looks good doing it.

WHAT IT DOES TO YOUR AUDIO

Nothing, unless you ask it to. DeltaScope passes the signal through unchanged and reports zero latency to your DAW. Once it's licensed, the only thing that can change what you hear is listening to the delta or the sum.

Setting or mode	What reaches the output
`Delta` mode, with `Hear the Delta` on (the default)	The difference signal: what a processor is changing.
`Sum` mode, with `Hear the Sum` on (off by default)	The two signals added together.
Any other comparison mode	The unchanged track.
The monitor switches off, in Settings > General	The unchanged track.

Setting or mode	What reaches the output
An offline bounce	The unchanged track, whatever the monitor switches say.
Gain and the band filters	No change: they affect only what's drawn, never what passes through.
The alignment offsets	They move the traces on screen, and change the delta or sum you hear, which is taken after alignment.

With any sidechain, `Hear the Sum` and `Hear the Delta` play 5 ms behind the live track, so that they can fade through silence when you switch them. With another DeltaScope instance as the sidechain they can play further behind still (see Hear the Delta). The track itself still passes through with no latency. (Without a licence, demo mode also mutes the audio briefly at intervals; see Demo mode.)

FEATURES

- **Delta**: subtract a reference from the track and see what differs; identical signals draw a flat line
- **Hear the Delta**: listen to the difference signal on its own, isolated in your monitors
- **Hear the Sum**: listen to the two signals added together, so Sum and Delta become a one-click polarity check
- **Magnitude view**: the live frequency-response curve of whatever processing is between the two signals, drawn from your own material; no test signal needed
- **Instance linking**: compare against another DeltaScope in the project (same plugin format), picked by track name; no routing needed
- **Sidechain input**: route a reference into it, the same way you route a key signal to a sidechain compressor
- **Comparison modes**: Overlay, Sum, Delta, Σ+Δ and Stack, against either kind of reference
- **Alignment**: offset either signal in samples or ms, to single-sample precision, or type a chain's latencies as a sum; ADC (auto delay compensation) measures a chain's latency for you, on either signal row, or, across linked instances, by reading the first sound after play or listening to the music
- **Multi FFT**: multiresolution analysis that resolves sub-bass detail without slowing the rest of the spectrum
- **Spectrum analyser**: FFT from 2048 to 65536 points, peak hold, `Slope`, crosshair with note and level readouts
- **Distortion readout**: feed it a test tone and read its THD, THD+N and IMD, with the noise floor and the level of each harmonic, beside the peak in the spectrum
- **Phase correlation**: the track against the reference, shown along the waveform, so you see where they are in phase and where they cancel
- **Band filters**: zero-phase high-pass and low-pass on main and sidechain, linkable
- **Level meters**: sample or true (inter-sample) peak, momentary, short-term and integrated LUFS (ITU-R BS.1770), RMS, crest factor
- **Mid-Side**: Mid (sum) and Side (difference) alongside Left, Right, Mono and L/R; either pair in two lanes, or in one lane with the second channel in front
- **MIDI note-lock**: send it the note you're playing and the window sizes itself to that pitch and holds the wave still; synth tuning at any pitch, without touching a slider
- **Freeze**: hold the scope, the analyser and the meters at one moment while the audio keeps playing, then zoom, pan and measure it, or change mode or timebase on it
- **Zoom and measurement**: zoom to 64×, pan, and click-drag measurement with frequency, note and level readouts
- **Colour by Frequency**: tint the waveform and the spectrum by what they contain, red in the bass to violet at the top, set for each channel in the colour picker
- **Naked mode**: hide every control and border for a clean overlay in videos, tutorials and livestreams
- **Export**: high-resolution image export, and settings export/import for moving between machines
- **Everything in our free oscilloscope, Scope**: beat-synced and Free-run display, clip indicators, Rolling mode, resizable to 4096 x 2160

SYSTEM REQUIREMENTS

Requirement	Minimum
OS	macOS 10.15 (Catalina) or later, or Windows 10 or later
Mac	Any Intel or Apple silicon Mac that runs macOS 10.15
PC	Any 64-bit PC that runs Windows 10
GPU	Any GPU with OpenGL 3.2 support
Formats	VST3, AU, CLAP, AAX (AU is macOS-only)

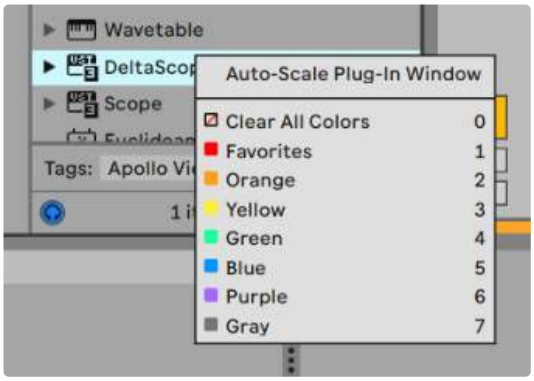
ABLETON LIVE ON WINDOWS

On Windows, Live enlarges every plugin window by your desktop's scale factor. That's Live's `Auto-Scale Plug-In Window` setting, and Live ships with it ticked (on) for every plugin. DeltaScope draws at the screen's real resolution, so Live's enlargement only blurs it: for DeltaScope you want this setting unticked (off).

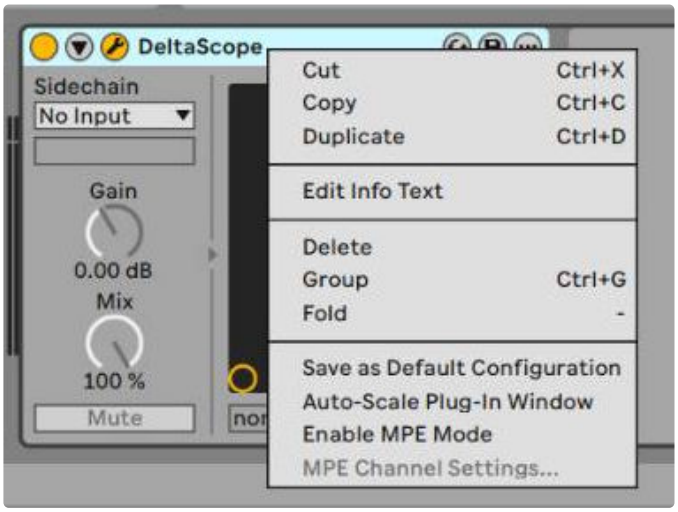
1. For every instance you add from then on: right-click the plugin in Live's browser and click `Auto-Scale Plug-In Window` so it's unticked.
2. For an instance already in your set: right-click the device's title bar and untick the same entry there.

If you're unsure, open the menu again: no tick beside the entry means Live is leaving the window alone.

The setting is per plugin, so if you use Scope as well, untick it for Scope too. Live also ticks it again whenever the plugin is updated, so expect to repeat this after installing a new version. If the display goes soft again, with a thin light line along the top of the scope, the setting is back on.



In Live's browser: right-click the plugin, untick `Auto-Scale Plug-In Window`.



Or on an instance already in the set: right-click the device title bar.

QUICK START

1. Install, open your DAW, insert DeltaScope on a track.
2. The first time you open it you'll see the activation screen. Enter your email and licence key and click `Activate`, or click `Continue in demo mode` to try everything first (see Demo mode).
3. Press play. The timebase starts in Sync at the 1/4 division, so one beat holds still against the grid.
4. DeltaScope opens slim: a scope on its own, the same shape as our free Scope. Make the window taller and the spectrum appears below the scope, with a gold pill between the two. Drag the pill to set how much of the display each pane gets; drag it to the bottom to show the scope alone, or to the top to show the spectrum alone.

That's all it takes to watch a track.

YOUR FIRST COMPARISON

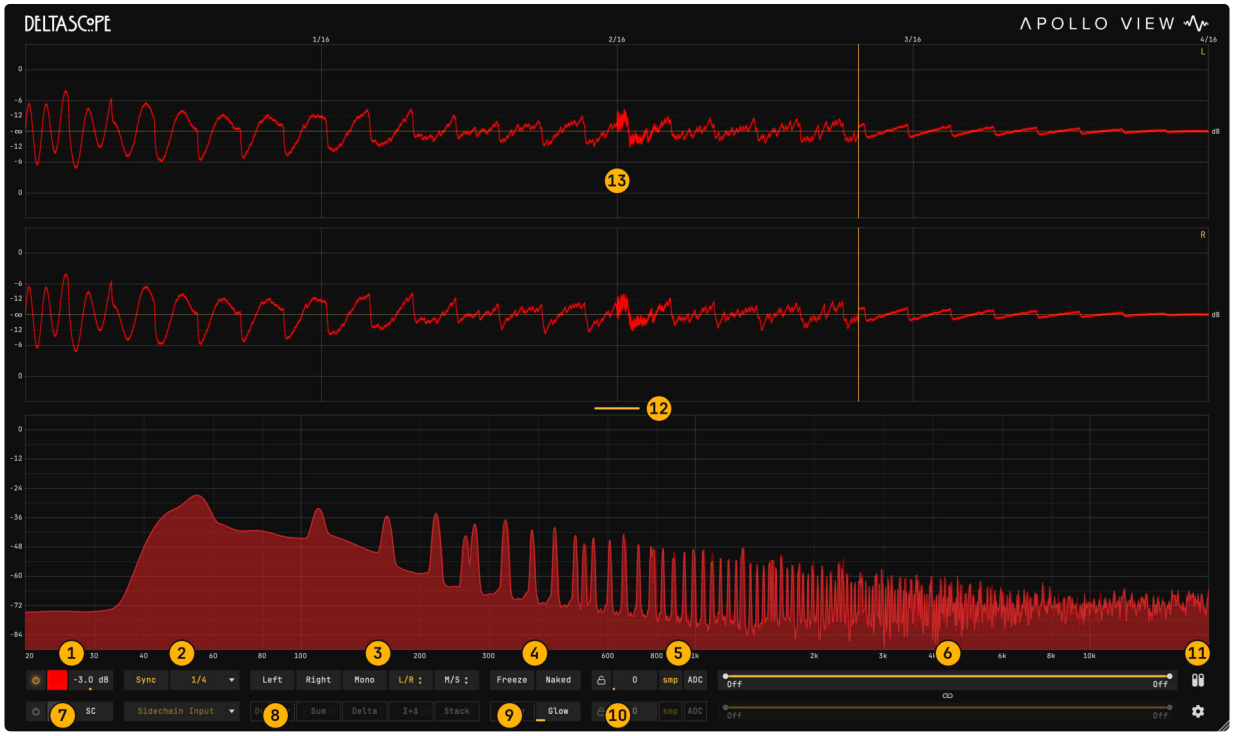
The delta is the difference between two signals, so it only means "what the processing changes" when the two are the same audio before and after that processing. Here's the simplest way to set that up:

1. On one track, insert an EQ with a clear boost or cut. Insert one DeltaScope **before** the EQ and another **after** it. No routing is needed.
2. Open the DeltaScope after the EQ and press `SC`. The source picker beside it is greyed out until SC is on. Open it and pick the other instance: it's listed under the track's name. (An instance never lists itself, so on this track the name means the one before the EQ.)
3. Press play. Once the transport has run, the two signals are lined up by their position in the project, so the DAW's delay compensation is handled for you. If the EQ itself adds latency (a linear-phase or lookahead mode), your DAW may already line the pair up: start with the offset at 0. In Ableton Live, type that latency into the offset, or let `ADC` measure it (Alignment lists which DAWs need what, under Latency between two linked instances).
4. Click `Delta`. The display now shows what the EQ changes and nothing else. **From this click the track's output changes too**: Hear the Delta is on by default, so what you hear is the EQ's change on its own, with everything the two signals share cancelled out.
5. Bypass the EQ and the two signals are identical: they cancel completely, a "null". The line is flat and the output silent (see The perfect null).
6. Click `Overlay` and the output is the unchanged track again. To look at the delta without hearing it, switch `Hear the Delta` off in Settings > General.

You can also compare against a track elsewhere in the project, or route a reference into DeltaScope's sidechain input the same way you'd route a key signal to a sidechain compressor, and pick `Sidechain Input`. The delta is then the difference between two separate signals, not the change made by one processor.

> **TIP:** The quickest way to learn the controls is to hover over them: with tooltips on, every control describes itself in the title bar. Below 1230 pixels wide the strip between the logos is too narrow for that, and shows no tooltips. The same strip tells you when `ADC` finds no reading, for ten seconds. Naked mode hides the bar, so it shows nothing there.

OVERVIEW



DeltaScope, numbered to match the key.

KEY

1 Main row: power, waveform colour (opens the colour picker), gain.
2 Timebase: Sync / Free / MIDI, with the division, window or octave beside it.
3 Channel modes: Left, Right, Mono, L/R, M/S.
4 Freeze and Naked.
5 Main alignment: padlock, sample offset, units, ADC.
6 Main band filter (HP/LP handles) and band link.
7 Sidechain row: power, colour (opens the colour picker), SC, source picker.
8 Comparison modes: Overlay, Sum, Delta, Σ+Δ, Stack.
9 Phase and Glow.
10 Sidechain alignment and band filter.
11 Meters button, with the settings cog under it.
12 The divider: the gold pill between the panes. Drag it to set how much of the display the scope and the spectrum each get, or drag it to either end to show one pane alone.
13 The display: scope lanes above, spectrum pane below.

The top row of controls belongs to the **main** signal, the bottom row to the **sidechain**. Everything above the controls is the display, and it's two panes: the scope, and the spectrum analyser under it. You can drag the divider between them (see Two panes, one divider).

RESIZING

Grab any edge or corner and resize. DeltaScope works from 1005 x 300 pixels up to 4096 x 2160, so a full-screen analyser fits on a 4K display; on anything larger than 4K the plugin window stops growing at 4096 x 2160. The size is remembered per instance, and `Save As Default` in Settings makes your preferred size the default for new ones. A new window still opens at 1230 pixels wide. Below 1230 the view buttons fold into menus (see Channel modes and Comparison modes), and the title bar shows no tooltips. At 1230 and wider you see no difference. Naked mode can go smaller still.

SLIDER GESTURES

Every value slider responds to the same gestures. Drag it to adjust: the faster the drag, the more the value changes. A plain click never jumps the value to the click point; it lets you type an exact value instead (Enter commits, Esc cancels). Alt-click resets it to its default. The mouse wheel also adjusts it: slow wheel clicks change the value in fixed steps; a fast spin moves it a long way across the range. The two-handle band sliders are drag-only. On Windows, Alt is the Alt key; on a Mac it's the key marked Option. The tooltips say Alt on both.

SETTINGS AT A GLANCE

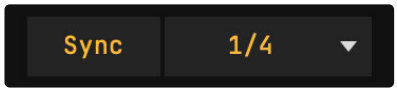
The cog at the bottom right opens Settings over the display. The pages are listed down the left (`General` , `Scope` , `Phase` , `FFT` , `Meters` and `About`), and clicking a name opens that page. Every change applies as you make it. Close it with the × at the top right, or with Escape. The Phase, FFT and Meters pages are described in their own chapters. Saving defaults, moving settings between machines and exporting the display as an image are in the Settings chapter.



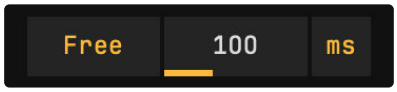
Settings, open on the General page: the pages listed down the left, the × at the top right closes it.

TIMEBASE

The timebase decides how much time the screen spans, and what the display is locked to, if anything. The mode button cycles `Sync`, `Free` and `MIDI`.



Sync: the beat-division menu.



Free: the window slider and its units button.



MIDI: the octave slider.

SYNC

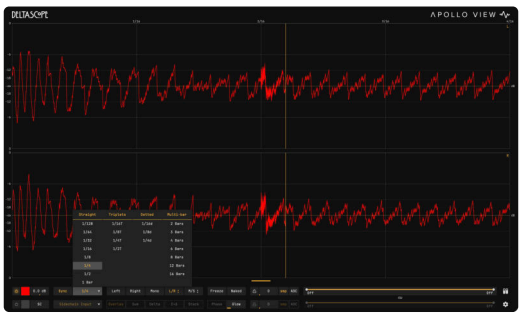
Locks the display to the host's tempo and transport, so a beat or a bar stays in the same place against the grid, and the same hit is drawn in the same place on every repeat. The beat-division menu sets how much time the screen spans: short divisions to study the shape of a single hit, 1/4 for a beat, multi-bar spans to watch dynamics across a phrase.

Group	Divisions
Straight	1/128, 1/64, 1/32, 1/16, 1/8, 1/4 (default), 1/2, 1 Bar
Triplets	1/16T, 1/8T, 1/4T, 1/2T
Dotted	1/16d, 1/8d, 1/4d
Multi-bar	2 Bars, 3 Bars, 4 Bars, 6 Bars, 8 Bars, 12 Bars, 16 Bars

Every instance reads the same host position, in every timebase mode. That's why traces on different tracks line up on the same grid, once any plugin latency ahead of them is compensated (see Alignment). It's also what lets a linked instance (see Sidechain) align its audio with yours once the transport has run.

DeltaScope can hold up to the last 64 seconds of audio at every sample rate up to 192 kHz, which is 16 bars at 60 BPM. At a slower tempo a window longer than that shows its most recent 64 seconds, and the title bar says so while you aren't pointing at a control: "Window limited to 64 s at this tempo (about 13 bars)." Above 192 kHz the bar lengths that can't fit are greyed out in the menu.

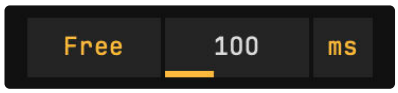
It doesn't keep all 64 seconds at all times, though: to save memory, each instance holds on to what its window needs, and never less than about 20 seconds. So when you switch to a window longer than that, the display fills forward from what it kept instead of showing the past at once.



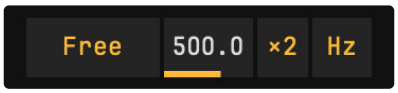
The beat-division menu.

FREE-RUN

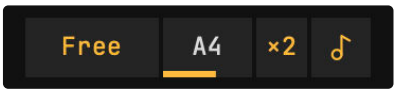
Free-run unlocks the display from the tempo, for audio with no transport or tempo to lock to: a synth you're playing by hand, hardware running on its own clock, a sample you're auditioning.



Free in time units: the window length and the units button.



Free in frequency units: the frequency, the cycle button and the units button.

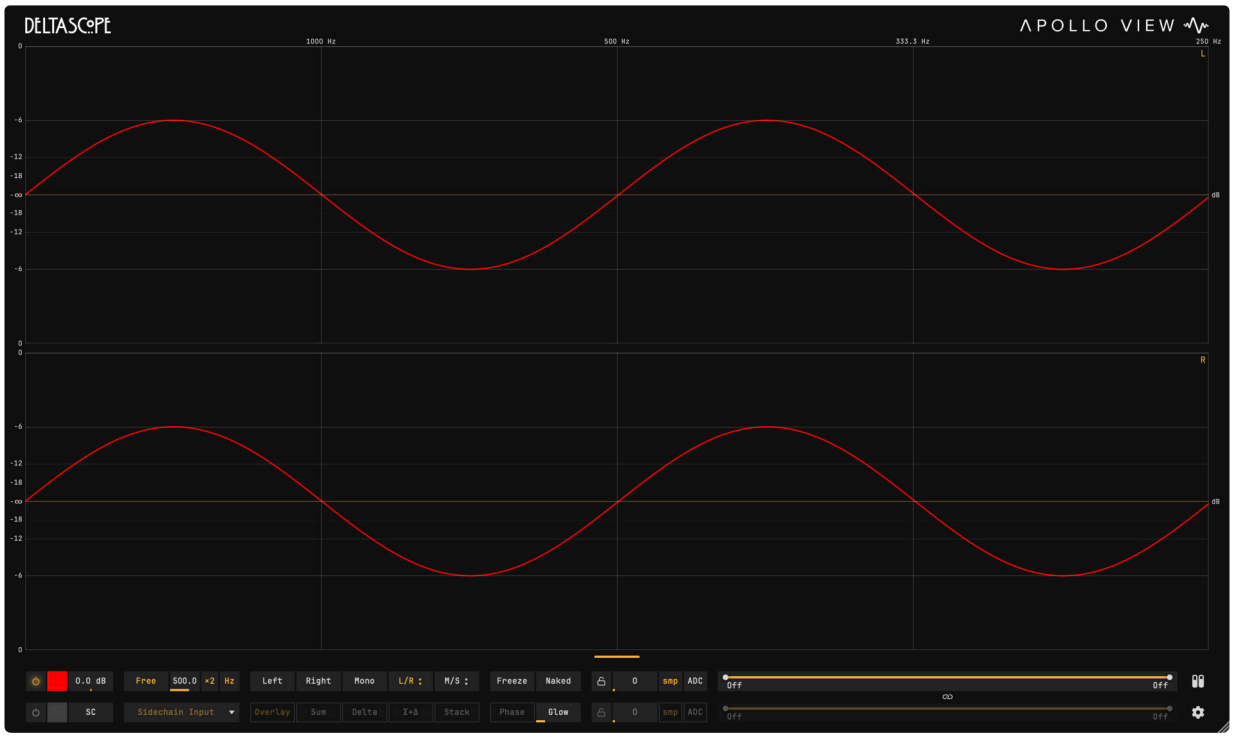


Free in note units: the note, the cycle button and the units button, now showing the note glyph.

- **Window**: in time units the slider sets how much time the screen spans, 10 ms to 10 s. (In frequency and note units the window follows the tone instead, and can be far shorter than 10 ms.) Click it and type "250", "250 ms" or "1.5 s", and Alt-click returns it to 500 ms. From one second up the readout switches to seconds: "1.50" is 1.5 s.
- **Units**: the gold button at the end of the slider changes what the slider sets: a length of time, a frequency or a note. Click it to cycle through the three. It always shows the unit you're in: `ms` or `s`, `Hz` or `kHz`, or a note glyph. It stays gold in every mode because it's a button, not a label.
- **Frequency**: dial or type a tone's frequency, up to 10 kHz (you can type "2k"; from 1000 Hz the readout is in kHz), and the window holds a whole number of its cycles. Alt-click resets to 500 Hz. The lowest frequency depends on the cycle count, because the whole window can never be longer than ten seconds: 0.1 Hz at ×1, 0.2 Hz at ×2, 0.4 Hz at ×4.
- **Note**: the same idea with the arithmetic done for you: for a bass line in E you pick E1 instead of working out that it's 41.2 Hz. Pick the note, C0 to C8 a semitone at a time (A4 is 440 Hz). Type a note name, "C#2" or "Db2", or type a frequency and the nearest note is chosen. Alt-click resets to A4. The grid labels stay in Hz.
- **Cycles**: in frequency and note units, the button between the value and the units button sets how many cycles of the tone fill the screen: `×1`, `×2` or `×4`. Two by default; Alt-click returns it to ×2.

> **PRO TIP:** Two cycles are easier to read than one. A free-running sweep starts at whatever point of the wave the audio has reached: with one cycle on the display, the shape's recognisable features are split between the two edges, and there's no repeat to compare against. That's why the cycle button defaults to `×2`. The MIDI octave slider is the same idea, driven by the played note instead of a value you set: MIDI starts at one cycle, and setting its octave slider to −1 shows two (see MIDI).

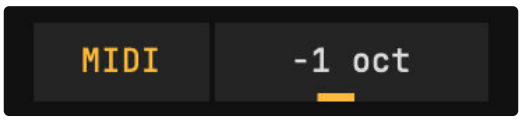
Free-run makes no attempt to hold the waveform still; each sweep draws the latest stretch of audio. A real tone is rarely at exactly the frequency you dialled in, so the stretch on screen is rarely a whole number of the signal's actual cycles. Each sweep then starts at a slightly different point in the cycle, and a steady tone drifts slowly across the screen. Holding it still is what MIDI note-lock is for.



Free-run in Hz units at the ×2 default: the frequency set to 500 Hz for a 500 Hz tone, and two whole cycles filling the screen.

MIDI

`MIDI` locks the window to the note you're playing. Send MIDI to DeltaScope's input and the display sizes its window to one cycle of that note's pitch; the octave slider widens or narrows the window in whole octaves, −4 to +4. At −1 the window is sized for the note an octave below, so two cycles of the played note fit; at +1, half a cycle.



The octave slider at −1: two cycles of the held note.

This is the mode for synth tuning: hold a note and see its actual cycle shape at any pitch, without adjusting any slider.

MIDI setup by DAW

Getting notes into DeltaScope usually takes one routing step. In most hosts the notes on an instrument track go to the synth and stop there; the effects after it never receive them.

REAPER: REAPER passes MIDI along the FX chain, so no extra routing is needed.

1. Insert DeltaScope on the instrument's track, after the synth. It already receives the notes.
2. To send it notes from a different track instead, add a send from that track to DeltaScope's track and enable MIDI on the send.
3. Play a note and click the mode button until it reads `MIDI`.

Ableton Live: the notes have to be sent to DeltaScope from a MIDI track.

1. Create a MIDI track carrying the same notes as the instrument: duplicate the clip onto it, or point the instrument track's `MIDI From` at the new track so one clip plays both.
2. On the new MIDI track, set `MIDI To` to the audio track hosting DeltaScope, then choose DeltaScope in the second chooser beneath it.
3. Play and click DeltaScope's mode button until it reads `MIDI`.

Logic Pro: Logic sends notes to the plugin in a track's instrument slot, never to one in the Audio FX chain. So for the note-lock DeltaScope goes in an instrument slot, on a track of its own, and takes its audio from Logic's `Side Chain` menu.

1. Create a Software Instrument track. Click its instrument slot and choose `AU MIDI-controlled Effects`, then Apollo View, then DeltaScope, then Stereo.
2. At the top right of the plugin window, set Logic's `Side Chain` menu to the track you want to look at. That track's audio arrives as DeltaScope's main signal, not on its `SC` row, and only while Logic is playing.
3. DeltaScope passes that audio on, so the new track plays the source a second time. Pull the new track's fader all the way down.
4. Click DeltaScope's mode button until it reads `MIDI`.
5. Press play and send notes to the new track: select it and play your keyboard, or copy the synth's MIDI region onto it. Let go of the note and the picture keeps that note's window until the next one.

If you'd rather not add a track in Logic, use Free-run's note units: set the timebase to `Free`, click the units button until it shows the note glyph and pick the note you're playing. At `×2` two cycles fill the screen, as the note-lock shows them with its octave slider at −1. Without the lock the picture drifts slowly rather than standing still, but there's nothing to route and nothing to work out.

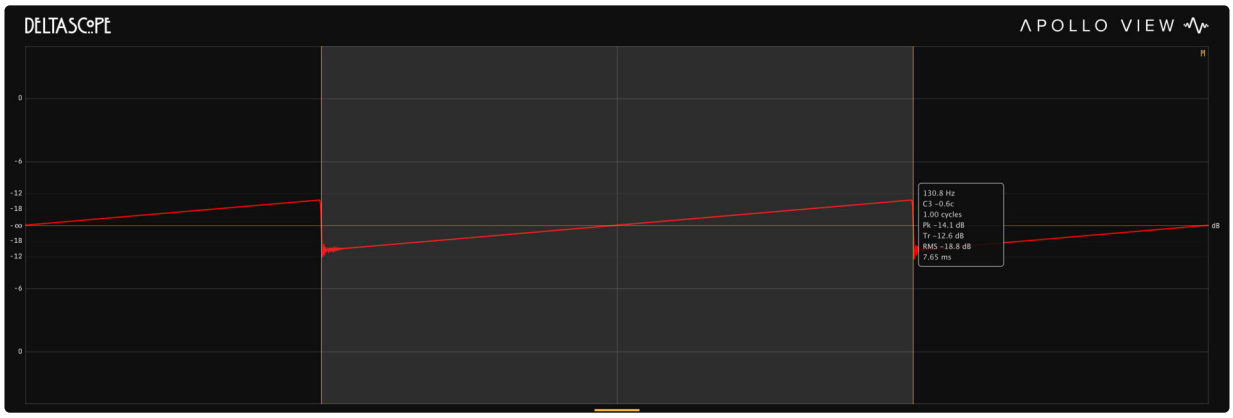
Other hosts: anything that can route a MIDI track's output to an effect plugin can send notes to DeltaScope's MIDI input. Where a host sends MIDI only to instruments and doesn't offer DeltaScope as one, use Free-run in note units with the note picked by hand, as described for Logic.



A held C3 saw, note-locked: the window is sized from the note's pitch and snapped to the same point in the wave on every redraw. The picture stays motionless, even from a source that drifts in pitch.



The octave slider at −1: the same locked note with two whole cycles framed, the MIDI-mode version of ×2.



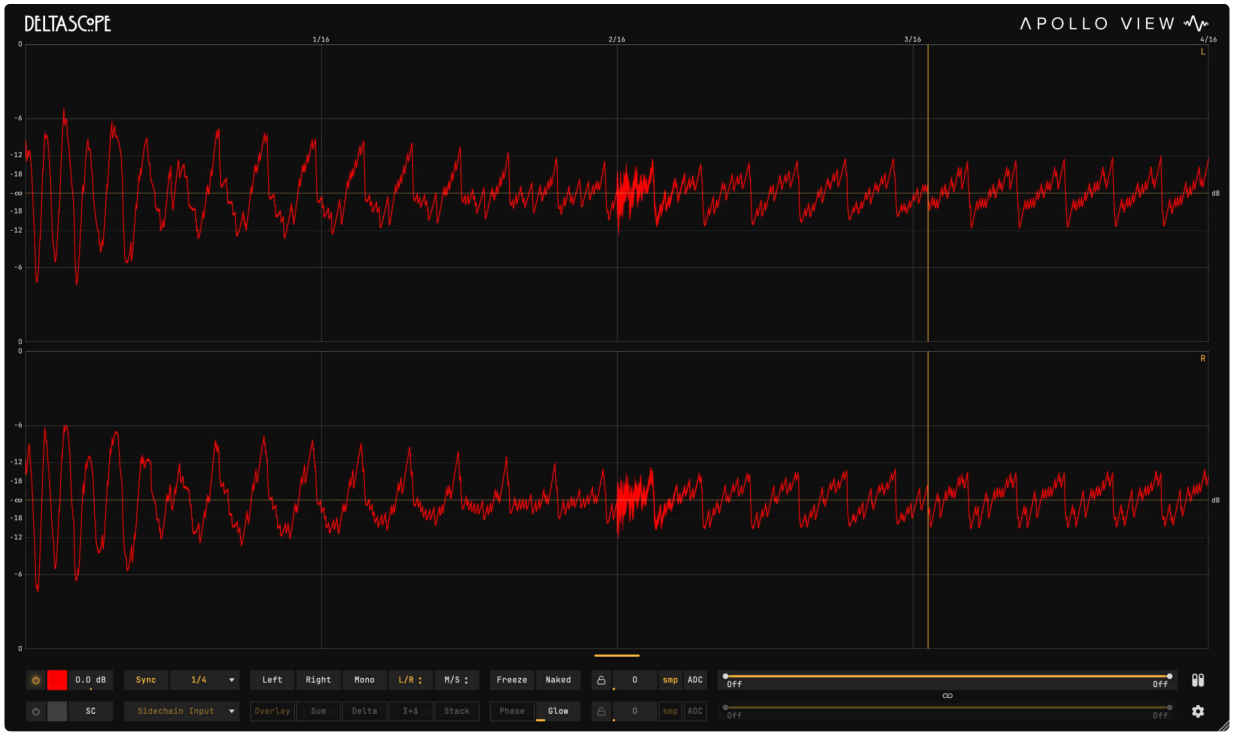
A locked wave can be measured like a frozen one (see Zoom, pan and measurement): drag across a cycle for its frequency, nearest note and level. The edges snap to the *fundamental's* zero crossings; on a saw that's the reset itself, so the selection runs from one reset to the next.

On every redraw, DeltaScope finds the same rising zero crossing of the wave (the same upward crossing of the centre line) and places it at the same position on screen, to within a fraction of a sample. It ignores the small variations that noise adds around that point, so it's far less likely to pick the wrong crossing. That's why a note stays motionless on screen even from an analogue source that drifts in pitch.

If the display glitches after you change MIDI routing, DeltaScope may still be holding the last note it received. Play a fresh note and it locks again.

The octave slider moves one octave per unhurried wheel click (a fast spin accelerates through more), and dragging it steps through the octaves with no acceleration.

THE DISPLAY



L/R lanes in Sync at 1/4: one beat against the 1/16 grid, with the cursor marking where the newest audio is being drawn.

DeltaScope draws the incoming waveform against a dB-calibrated grid:

- **Amplitude labels**: the dB levels of each lane, down to the minus-infinity centre line.
- **Time labels**: beats in Sync. In Free-run's time units, the elapsed time; in Hz or note units, the frequency of the tone whose cycle lasts that long. No time labels in MIDI: the window is one cycle of whatever note you play, so its length changes with the pitch.
- **Cursor**: the thin vertical line marks where the newest audio is being drawn. Switch it off with `Cursor` in Settings > Scope if you'd rather have a clean frame.
- **Lanes**: L/R splits into L and R lanes and M/S into Mid and Side, or either pair shares one lane (see Channel modes), and the comparison modes add or merge lanes (see Sidechain).

Turn the display gain up or down and the lines and labels move with it, so a label always tells you the real level of the audio, not the boosted picture.

TWO PANES, ONE DIVIDER

The display is two panes: the scope on top, the spectrum analyser below. You decide how much of the display each one gets. The gold pill on the line between them is a handle: drag it and the split follows your mouse. Near the middle the pill holds for a moment on an exact even split, which is also where it sits on a new instance once the window is tall enough for both panes; keep dragging and it follows the mouse again.

- The spectrum can be a small share under a large scope, the two can share the display evenly, or the spectrum can take most of the window with a short scope above it.

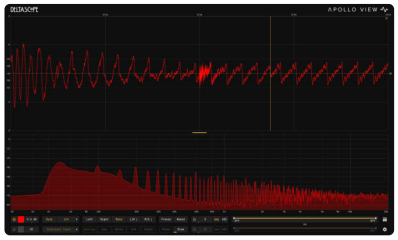
- Each pane has a minimum height: drag past it and that pane closes completely, with a short animation.
- Push the pill to the bottom and you have the scope alone; push it to the top and you have the spectrum alone.
- The pill stays where you left it. Drag it back to reopen the other pane.



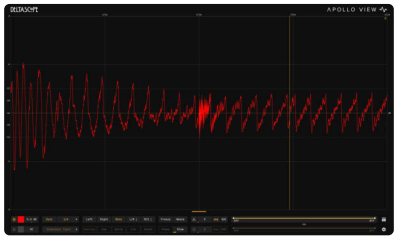
The divider pill, on the line between the two panes.

A new instance opens slim on purpose: a scope strip, the right height to leave open on many busy tracks at once, with the spectrum closed. Make the window taller and the spectrum opens beneath the scope as soon as there is room for both panes. You can also stay slim and swap panes instead: drag the pill to the top and the analyser takes the strip in place of the scope. The split is saved with your project per instance, and `Save As Default` in Settings makes your preferred layout the starting point for new ones.

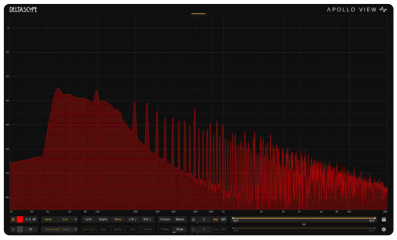
Naked mode hides the pill and keeps whatever split you had. Move the mouse near the middle of the line between the panes and a faint pill comes back, so you can drag the split without leaving Naked mode. It goes again a moment after you move away.



Both panes.



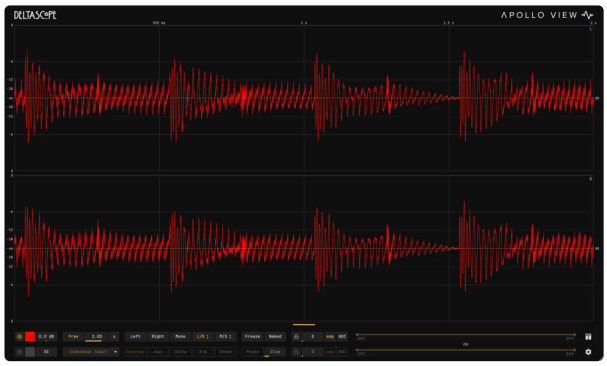
Scope only: pill dragged to the bottom.



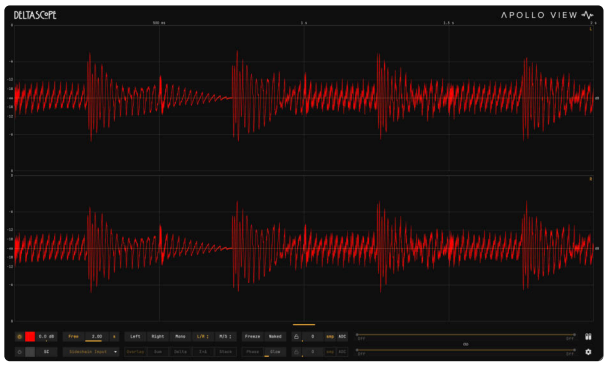
Spectrum only: pill dragged to the top.

ROLLING MODE

Rolling mode scrolls the waveform instead of redrawing it in place: the newest audio enters at the right edge and everything slides left, the way DJ software scrolls a waveform. The display still spans your chosen division, time window or note cycle; only the way the audio moves through it changes. To use it, enable `Rolling` in Settings > Scope. Zoom, the cursor, click-drag measurement, the band filters, Glow, the phase correlation display and the MIDI note-lock are inactive while rolling. Freeze a rolling display and measurement works again on the still waveform.



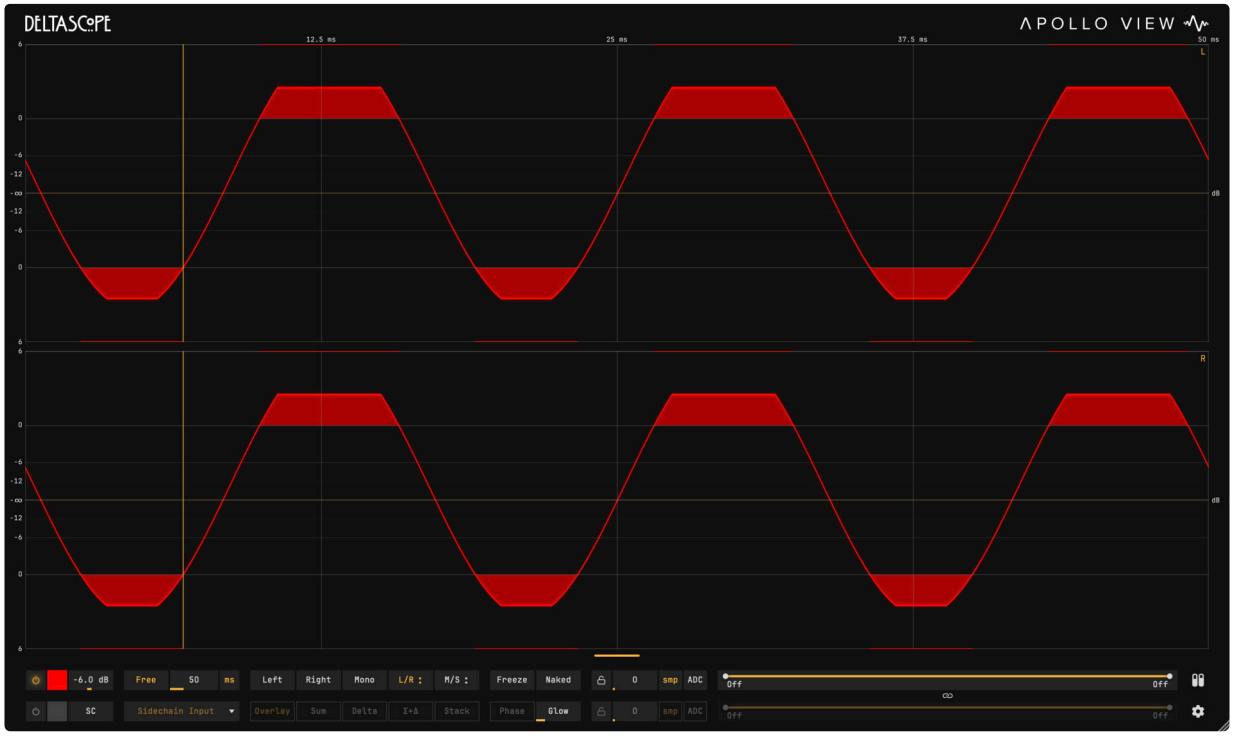
Rolling mode, Free-run at a 2 s window: the newest audio enters at the right edge…



…and a moment later everything has slid to the left.

CLIP INDICATORS

The display edge lights red where samples rise above 0 dBFS: top border for positive clipping, bottom for negative. Samples above 0 dBFS can happen inside a DAW, after a gain stage set too high, for instance. The mixer itself has headroom above 0 dBFS, but your converters and any fixed-point export don't. The clipped stretch of waveform is also drawn highlighted, so it stays visible in place. The border flash and the clip highlight each have their own switch in Settings > Scope: `Clip Border` and `Clip Overlay`.



A signal driven above 0 dBFS: border flash at the edges plus the highlighted clip overlay. Display gain at −6 dB keeps the waveform away from the border.

POWER, GAIN AND COLOUR

POWER

The power button switches the main signal's display on and off, both the scope trace and the spectrum curve. It never changes the audio. The sidechain row's power button does the same for the reference. Power buttons are for switching a signal's picture off while you work. They're never saved with the project: reopen it and every trace is on.



Power, colour swatch and the gain slider.

GAIN

Scales the displayed amplitude, −36 to +36 dB in 0.1 dB steps. It changes only the picture, never the audio. A new instance starts at −3 dB on purpose: many instrument plugins output a signal close to full scale. With 3 dB of headroom, the 0 dB line sits 3 dB inside the border, so a loud signal's peaks are still drawn as a shape instead of a flat line along the top and bottom borders. Alt-click returns it to −3 dB; typing and the wheel behave as they do on every other slider (see Slider gestures).

> **PRO TIP:** Look for overs before you touch the gain. At the −3 dB starting point the border sits about 3 dB above 0 dBFS, so peaks a little over 0 dBFS still draw as a shape, between the 0 dB line and the border, and the border lights red where it happens (see Clip indicators). Once the level is under control, raise the gain until the waveform fills the lane; the labels follow, so what you read is still the true level.

COLOUR

Click the swatch to recolour the main waveform. The sidechain has its own swatch in the row below. Each picker has two tabs, `L (M)` and `R (S)`, and an `Auto` button beside them.

The two tabs

- `L (M)` sets the signal's colour.
- `R (S)`: pick a colour here and the right channel (the side channel in Mid-Side) wears it instead, on the scope and in the analyser. The swatch then shows it as its right half.
- In the Right channel mode (see Channel modes) the one lane and its analyser curve wear the `R (S)` colour, with or without a sidechain showing, and so do the sidechain's.

Auto

`Auto` links the second channel back to the first. While it is lit the second channel follows the first in everything: its colour, and whether it is solid or by frequency. `R (S)` then shows the same choice as `L (M)`: one of `Solid` and `By frequency` is lit already, or neither is when the scope and the analyser are set differently (see When the channels or the displays differ).

`Auto` turns off in two ways:

- Pick a colour on `R (S)`. The second channel takes that colour.
- Choose a word on `R (S)` that is not lit: the other of `Solid` and `By frequency`, or either one when neither is lit.

Clicking the word that is already lit does nothing and leaves `Auto` on. When the second channel ends `Auto` by a choice of word, it starts from the lighter shade `Auto` was drawing for it where the two channels share a plot. That holds for the main swatch only: the sidechain's `Auto` colour is never split, so its second channel starts from the sidechain's colour as it is. Click `Auto` to link them again.

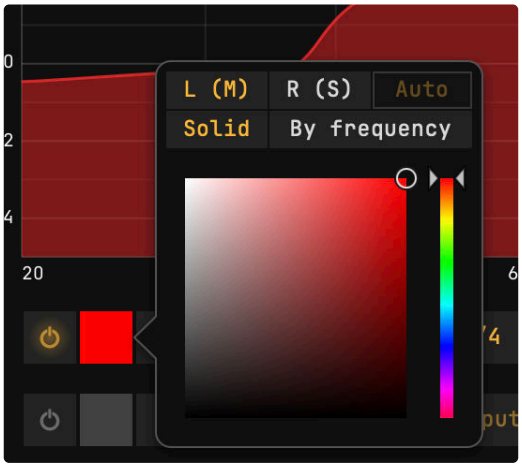
Solid and By frequency

Under the `L (M)`, `R (S)` and `Auto` buttons is a second row with two choices, `Solid` and `By frequency`. The one that is on is gold. The choice is for the channel whose tab is selected, so the left channel (or Mid) can be tinted by frequency while the right (or Side) keeps a plain colour, or the other way round.

- On `Solid` the picker shows the colour square and the hue strip.
- On `By frequency` it shows the frequency ramp in their place, with `Low` and `High` under it. The colour you had picked is kept and comes back when you choose `Solid`.

The choice sets that channel on the scope and in the analyser together (see Colour by frequency). Changing a colour or the choice does not move the analyser's zoom.

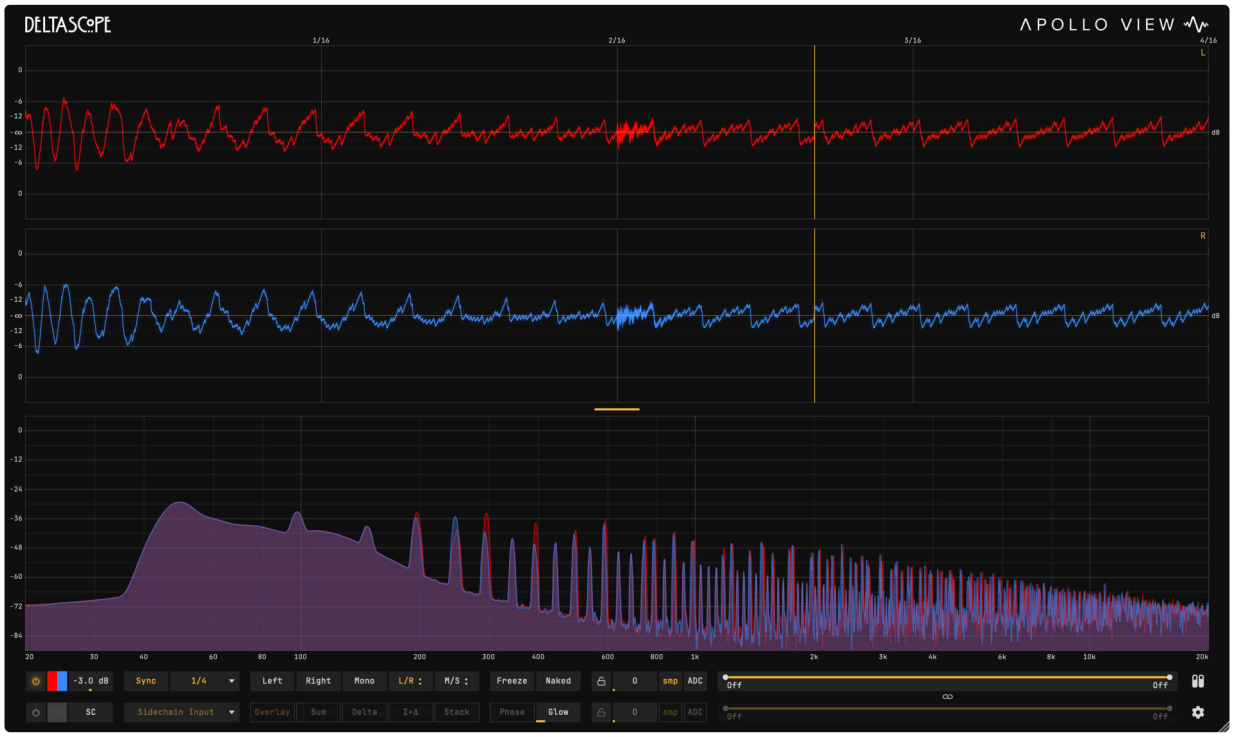
The analyser follows the same colours: each curve takes the colour of the signal it carries. With the second channel on `Auto`, the two curves in L/R or M/S sit either side of your colour, one a little darker and one a little lighter, so they stay apart. The scope's one-lane layout (see Channel modes) wears the same two shades, so a pair reads alike on the scope and in the analyser.



The colour picker, opened by clicking the swatch: the `L (M)` and `R (S)` tabs and `Auto` side by side, with `Solid` and `By frequency` under them, above the colour field.



The picker on `By frequency`: the ramp, with `Low` and `High` under it, replaces the colour square and hue strip.



A blue picked on the `R (S)` tab: the right channel wears it on the scope and in the analyser, and the swatch shows it as its right half.

COLOUR BY FREQUENCY

Colour by Frequency tints the waveform by what it contains instead of one flat colour. The colours run red through the low bass up to 80 Hz, orange and yellow above it, green through the mids, blue to violet at 20 kHz. It is the same ramp the analyser uses for its own Colour by Frequency (see Colour by frequency in the spectrum): switch both on and a note wears one colour on the waveform and on its peak alike.

Where several ranges sound at once the colours mix, so a kick reads orange at the hit, a hi-hat violet, and dense full-band material goes pale. It tells you where in the spectrum each part of the waveform lives without opening the analyser. The colour sits on the sound that made it: a

kick is orange from its first instant, and a hi-hat over a bass note shows violet and red where each sounds.

The newest edge
The newest 22 ms of a running trace (the edge beside the cursor, or the newest edge when rolling) is drawn from what has arrived so far. It settles over a frame or two, so a hit's first moments can change colour briefly just after they are drawn.

Where it is set
A signal can be coloured on the scope, on the analyser, on both, or on neither, depending on where you set it.

Where	What it sets	Shows on
The colour picker (see Colour)	`Solid` or `By frequency`, for each channel	Scope and analyser together
Settings > Scope > Colour by Frequency	A switch for `Main` and one for `Sidechain`	Scope only
Settings > FFT	Its own pair of switches, `Main` and `Sidechain`	Analyser only

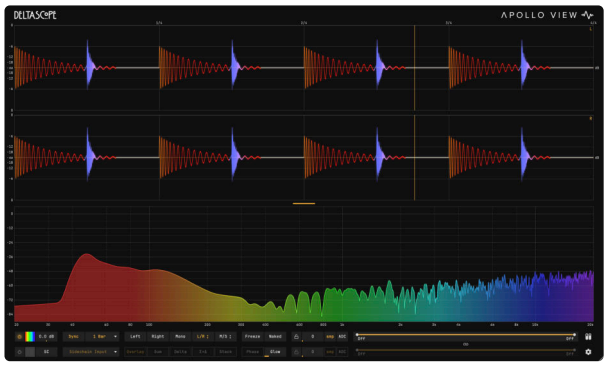
Each switch sets both channels of its signal on its display. Only the picker can make the two channels differ.

When the channels or the displays differ

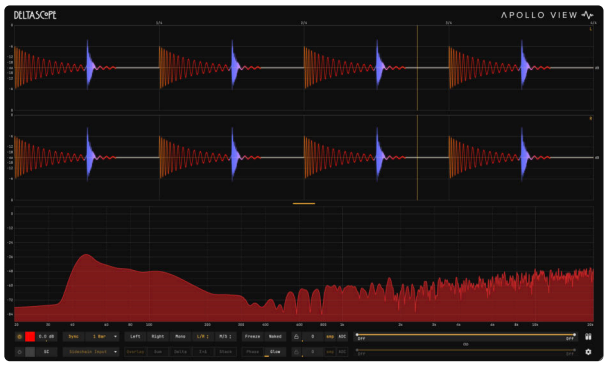
What you see	What it means
A Colour by Frequency switch with its knob in the middle and a small note, `L (M) only` or `R (S) only`	Only the channel in the note is by frequency. Click the switch and both channels turn on; `Reset` on the page turns both off.
In the picker, neither `Solid` nor `By frequency` is gold, and a line under them reads "Scope only" or "Analyser only"	The scope and the analyser are set differently for the selected channel, and the line names the display that is by frequency. Choosing either word then sets both.



A Colour by Frequency switch with its knob in the middle: only the left channel is by frequency, as the note says. Click it and both turn on.



A kick under hats, main signal only, Colour by Frequency on for the scope and the analyser: orange at each hit, violet in the hats, the analyser running red to violet along its axis, and the swatch showing the ramp.



The same audio with only the scope switch on: the waveform is coloured, the analyser wears the swatch's flat colour, and the swatch shows that colour.

The swatch

Click it to open the picker. The sidechain's swatch is greyed out, and opens nothing, until `SC` is on.

When	The swatch shows
Every channel it colours is by frequency on both the scope and the analyser	The ramp
Only one channel is	That channel's half as the ramp, the other half in the other channel's colour
Otherwise	The signal's colour, in two halves when a second colour is picked

The sidechain's swatch works the same way. A few cases are worth knowing:

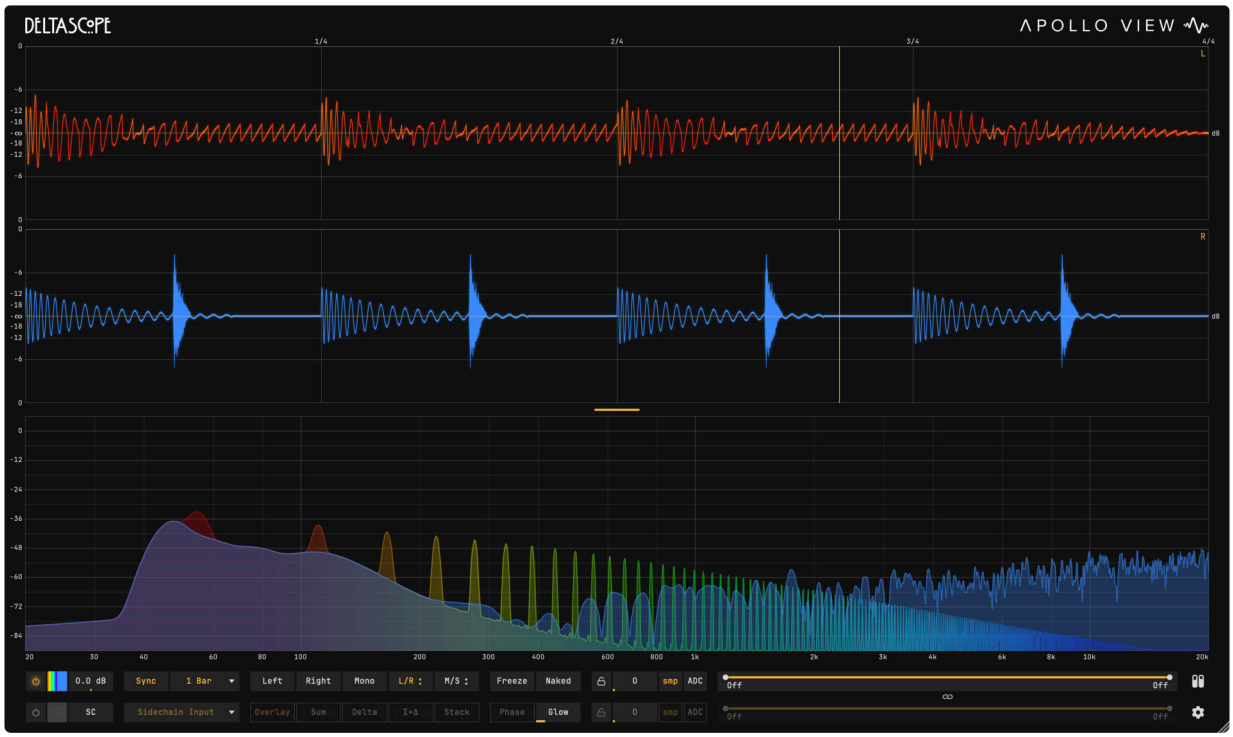
- **The sidechain in the analyser** follows the view. In Right its curve wears the `R (S)` tab's colour and choice, as its trace on the scope does. In L/R and M/S the analyser draws the sidechain as one curve, which wears the `L (M)` tab's.
- **The MIDI note-lock**: while it holds the trace still, the trace keeps the swatch colour.
- **A band filter**: a lane under an active band filter keeps colour by frequency and wears the colours of the band it shows (see Band filters).
- **A project from 1.3** opens as it looked: a switch that was on is on for both channels.



Only the left channel is by frequency: the swatch shows the ramp as its left half and the right channel's colour as its right half.



Every channel the swatch colours is by frequency: the swatch shows the ramp. Click it and the picker opens.



Left by frequency and right a plain blue, set in the picker: the same on the scope and in the analyser, and the swatch is half ramp, half blue.

CHANNEL MODES

Mode	Shows
`Left`	Left channel only.
`Right`	Right channel only.
`Mono`	The mono average, (L + R) / 2.
`L/R`	Left and Right, in two stacked lanes or in one. Default.
`M/S`	Mid (sum) and Side (difference), in two stacked lanes or in one.



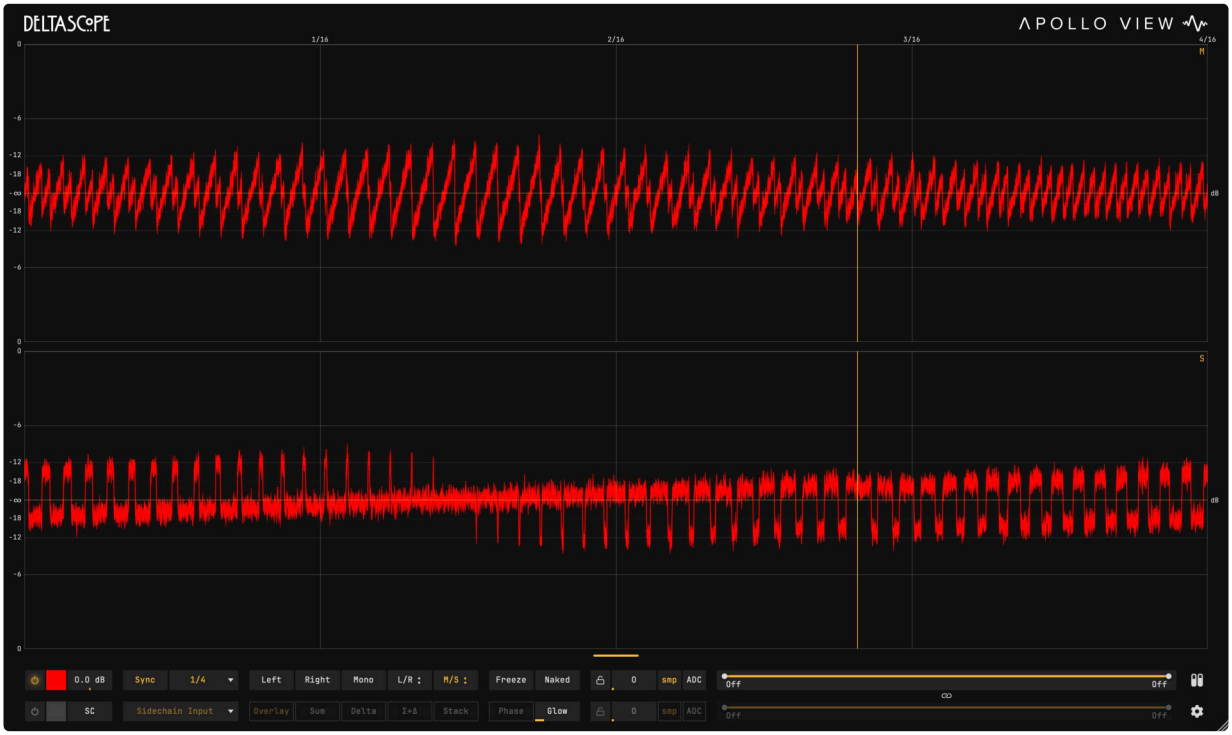
L/R selected, in two lanes.



M/S selected: the lanes become Mid and Side.

A mono track has only one channel, so DeltaScope feeds it to both lanes: L/R shows the same trace twice, and Mono, Left or Right give you a single lane of it.

M/S shows you the stereo image directly: anything that exists only in the Side lane disappears in mono playback. To check whether two elements are cancelling each other, route one in as main and the other as reference, then turn on the phase correlation display, which shows where they cancel.



M/S on a wide signal: a Side lane with a lot of signal means a wide image, and anything that appears only in the Side lane is lost in mono.

The low-end mono check that uses M/S is in Use Cases, under Check what survives mono.

Below 1230 pixels wide, the five buttons fold into one dropdown, and the sidechain row's comparison modes fold with them (see Comparison modes). Its menu lists `Left`, `Right`, `Mono`, `L/R` stacked, `L/R` overlay, `M/S` stacked and `M/S` overlay, and the dropdown shows

the current view with the mark of its layout. At 1230 and wider the buttons come back, with the view you picked lit. Nothing else in the row moves.



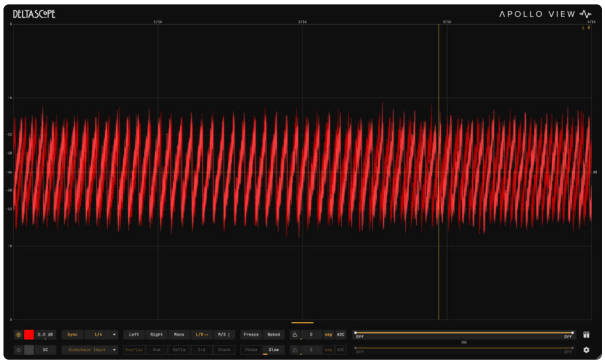
The folded view dropdown, open at the least window size, 1005 x 300: its seven items, with `L/R` stacked marked as the current view.

ONE LANE OR TWO

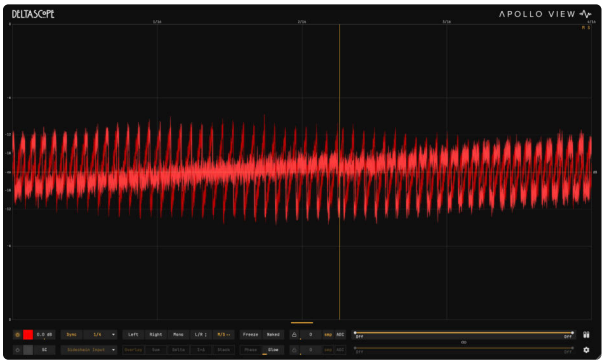
Click `L/R` again while it's the active mode and Left and Right share one lane at full height, Right drawn in front of Left; click once more and they're back in two lanes. `M/S` does the same for Mid and Side. Each button remembers its own choice and shows it as two dots: one above the other for two lanes, side by side for one.

In one lane the pair wears a darker and a lighter shade of the waveform colour, the same two the analyser's curves wear, or the colours you picked on the `L (M)` and `R (S)` tabs. One lane puts both channels on one grid, so a difference in level or timing between them is read straight off the picture; two lanes keep each channel's shape clear when the signal is dense.

With `SC` on, the pair goes back to its two lanes: a lane never holds more than two traces, and the reference takes one of them.



L/R in one lane: Right, in the lighter shade, in front of Left.



M/S in one lane: Side in front of Mid. A wide signal has a lot of Side.

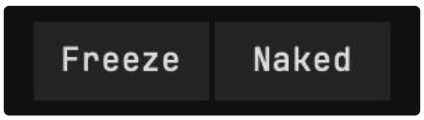
FREEZE AND NAKED MODE

FREEZE

`Freeze` holds the picture while the audio keeps playing, on every pane at once: the scope stops, the spectrum holds its curves, and the meters stop at the levels of that moment. The loudness readout shows the moment's momentary or short-term loudness; the integrated figure keeps the value it had. A frozen frame can be zoomed, panned and measured. Unfreeze and the live signal is back everywhere.

A freeze keeps the last 8 seconds, or the window on screen if that's longer, and the controls still work on it:

- **Channel mode, comparison mode, Delta view, FFT size**: switch any of them and you see that view of the frozen moment.
- **Colour by Frequency**: switch it on or off, or pick `Solid` or `By frequency` for a channel, and the frozen picture is redrawn that way.
- **Timebase**: the scope shows the same moment at the new one. Freeze on a bar, switch to a quarter note and zoom in on the last beat, or pick a longer window to see what led up to it (a window longer than 8 seconds shows as much as was kept).
- **The analyser and the meters**: open them later in a freeze and they show the frozen moment too.



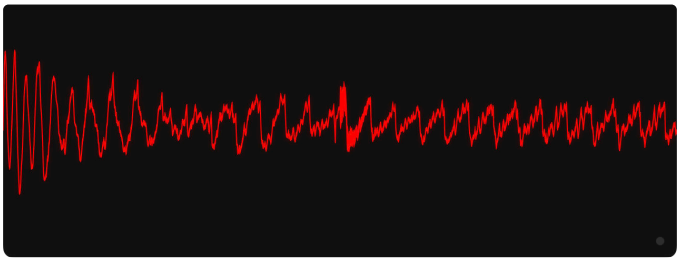
The Freeze and Naked buttons, both off.

Freeze is per instance and isn't saved with the project. A freeze stays held when you close the plugin window and open it again, but loading a project or preset, or changing the sample rate, brings the plugin up live, so a reopened project never opens with an old frozen picture.

NAKED MODE

`Naked` hides every control and border, and the display fills the whole plugin window: the waveform, the spectrum, or both, at whatever split you had. For a projector, a second screen, or just concentration. Naked mode isn't saved: close the plugin window and it opens with its controls again, and a project saved in Naked mode reopens in the normal view.

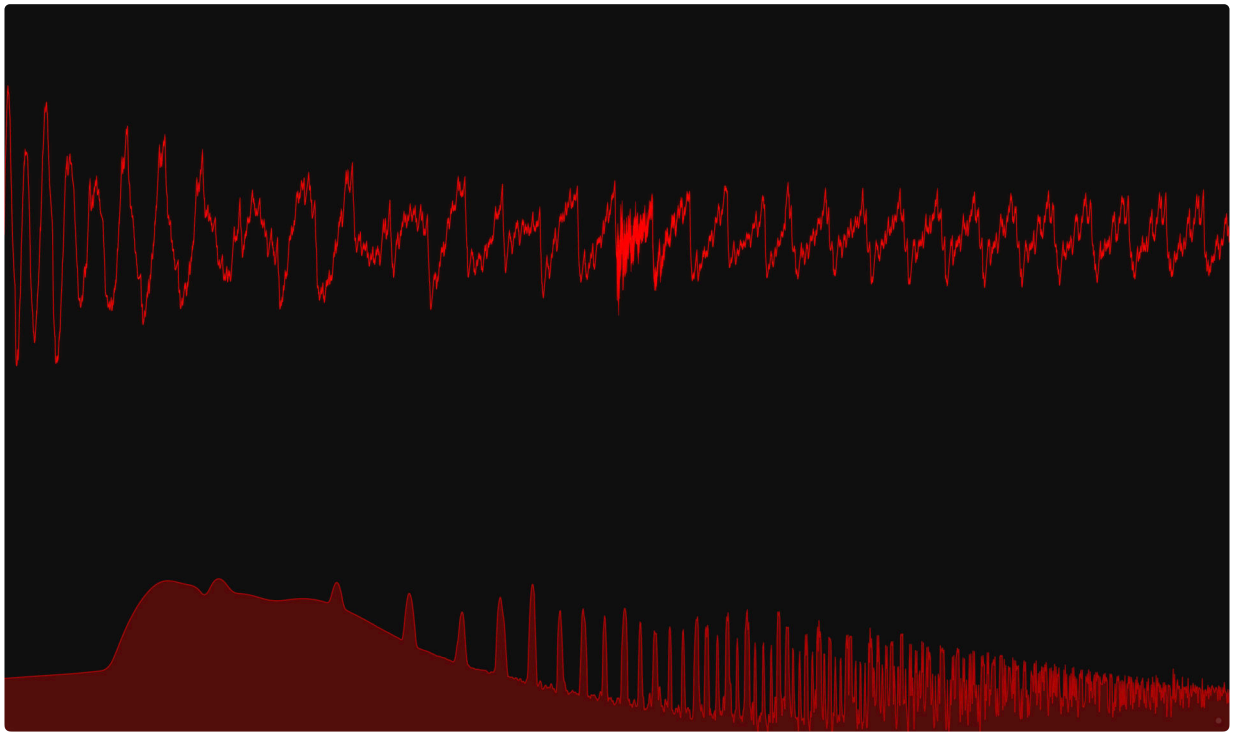
With the controls hidden the window can be made as small as 640 x 240. When you bring the controls back, a window smaller than 1005 x 300 grows to that size, with its top-left corner staying where it was.



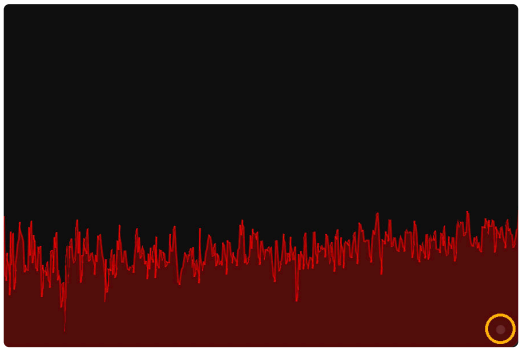
Naked mode at its least size, 640 x 240, with the restore dot in the bottom-right corner.

One dim dot remains in the **bottom-right corner**. Click it (tooltip: “Click to restore controls.”) to bring the controls back.

Naked mode works well together with Freeze, and it’s made for video. Put the plugin window in a screen recording, or use it as an overlay in a music video, tutorial or livestream.



Naked mode in Mono: waveform and spectrum edge to edge, ready for a video overlay.



Click the dot in the corner to restore the controls. In Naked mode it’s the only control left, apart from the faint divider pill that appears when the mouse is near the line between the panes.

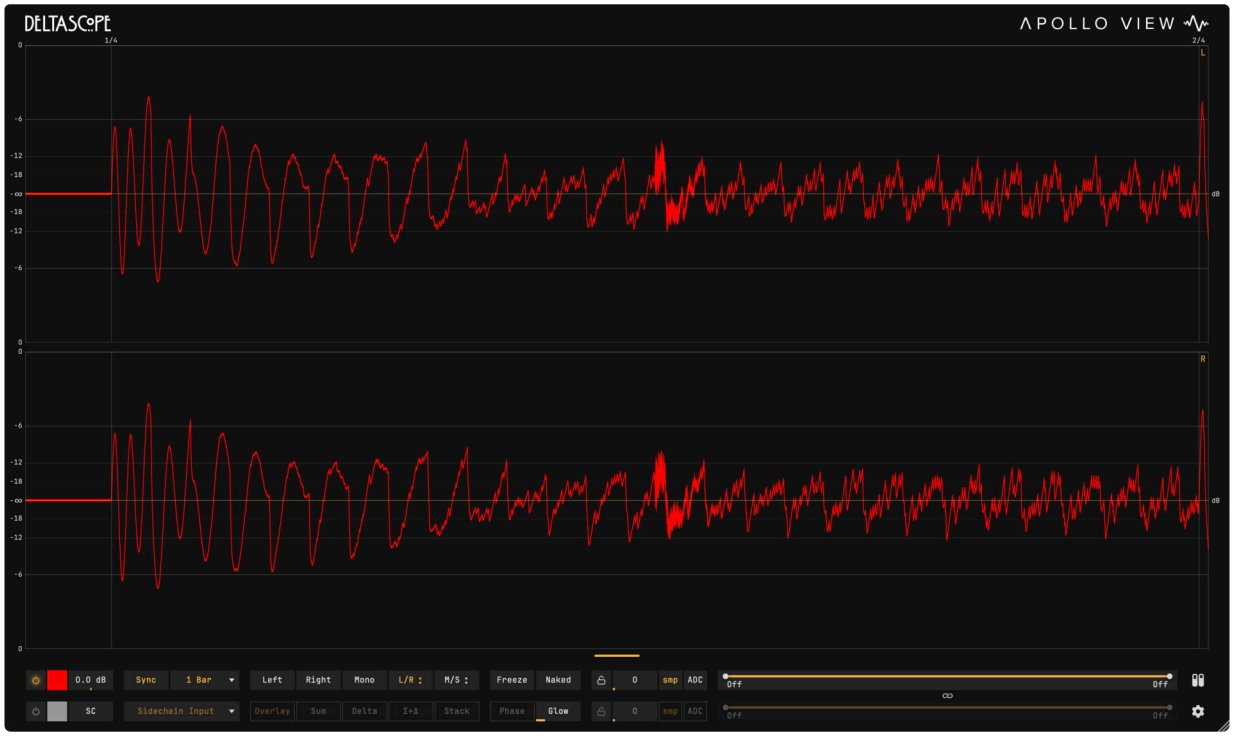
PRO TIP: There’s a hidden recording aid. Cmd-Alt-Shift-click the title bar (Ctrl-Alt-Shift on Windows) and the plugin window locks to 16:9 while you resize, so a capture fills a video frame edge to edge with no letterboxing. There’s no on-screen indicator; the only sign that the lock is on is that the window keeps 16:9 as you drag a corner. Click the same way to unlock. The lock lasts only while the plugin window is open and is never saved with the project.

ZOOM, PAN AND MEASUREMENT

ZOOM AND PAN

- **Scroll**: zooms the time axis towards the pointer, up to 64×.
- **Cmd-drag** (Ctrl-drag on Windows): pans left and right while zoomed.
- **Double-click**: resets to the full window.

Zoom changes only what you see, never the audio. It works on live or frozen waveforms, but not in Rolling mode. The same gestures work in the spectrum pane, which zooms up to 16× (the next chapter).

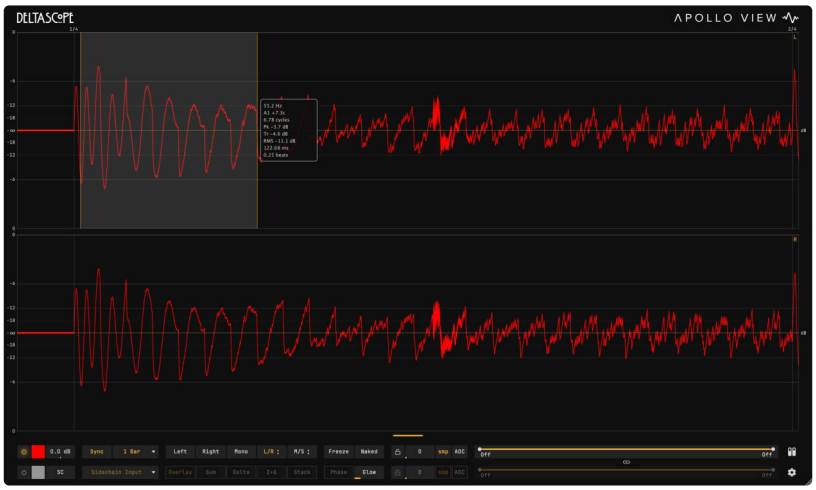


Zoomed into the note that starts on the second beat of a one-bar window; whatever is under your pointer stays in place as you zoom. Double-click returns to the full window.

MEASUREMENT

Click and drag across the waveform to measure a region. A selection stays on one side of the cursor line, since the audio either side of it is from different sweeps. The selection snaps its edges to zero crossings, and the readout shows:

- frequency of the selection (and the nearest MIDI note)
- peak, trough and RMS level in dB (Pk reads `--` when the selection never goes above zero, and Tr when it never goes below; RMS reads `--` when the selection is digital silence)
- duration in ms, in beats when the host reports a tempo, and the number of cycles

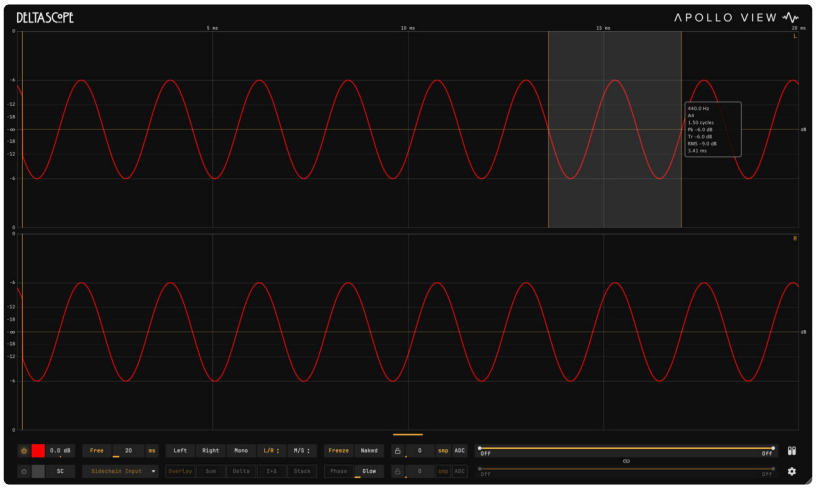


Zoomed in, drag across what you found. The selection snaps to zero crossings and reads its length and level, plus the frequency and nearest note of the repeating cycle found inside the selection.

Or hold **Alt** and hover. DeltaScope finds where the waveform repeats and selects one full cycle of the note under the pointer for you. It shows the same readout, accurate to within a couple of cents (hundredths of a semitone) on a clean, steady tone.

The selection stays on screen after you let go, until you click the lane again or press Esc: a right-click clears it, a left click-and-drag starts a fresh one. An Alt-hover selection clears on the next mouse move after you release Alt. (On a lane too short for the readout, a one-line hint appears instead: “Increase height to see measurements”. Make the window taller, or give the scope more of the split.)

> **TIP:** Measure on a still image. Press `Freeze`, take your time dragging the selection, then unfreeze. On a live display the waveform keeps redrawing under your selection; a looping beat in Sync, or a MIDI-locked note, holds still enough to measure live.

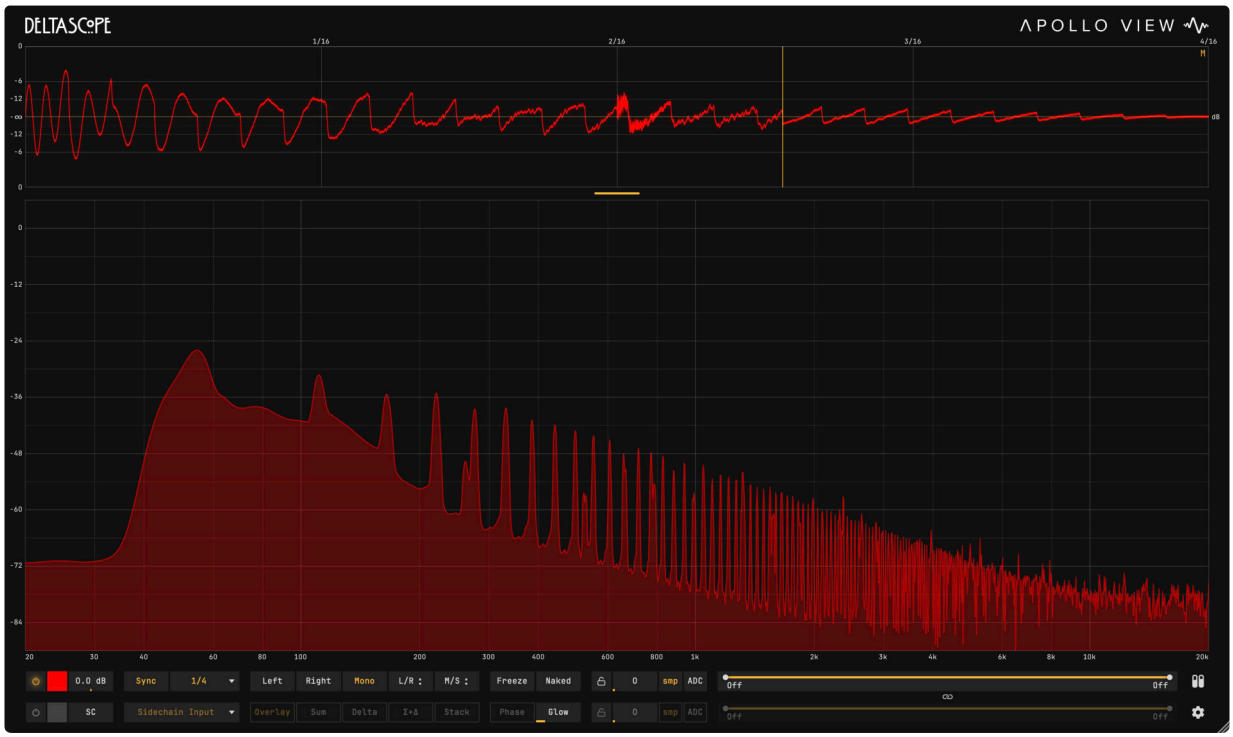


Drag-measuring a frozen 440 Hz sine: the edges snap to zero crossings, and the readout gives the frequency, the nearest note (A4), the length in cycles (1.50 here), the peak, trough and RMS levels, and the duration.

SPECTRUM ANALYSER

The scope shows when something happens; the spectrum shows which frequencies it is made of. They answer different questions about the same audio, so DeltaScope can keep both on one screen, or give the whole display to either one (the divider is described under The Display).

The analyser is an FFT: the maths that splits a stretch of audio into the frequencies it contains and shows the level of each one. Scroll to zoom, Cmd-drag to pan (Ctrl on Windows) and double-click to reset, as in the scope; the crosshair follows plain mouse movement. The zoom stays put while you change things in Settings > FFT; only a new `Min` or `Max` frequency returns the pane to the full range.



Spectrum pane opened wide (Mono): the live curve, tilted +3 dB/oct so a balanced mix reads flat.

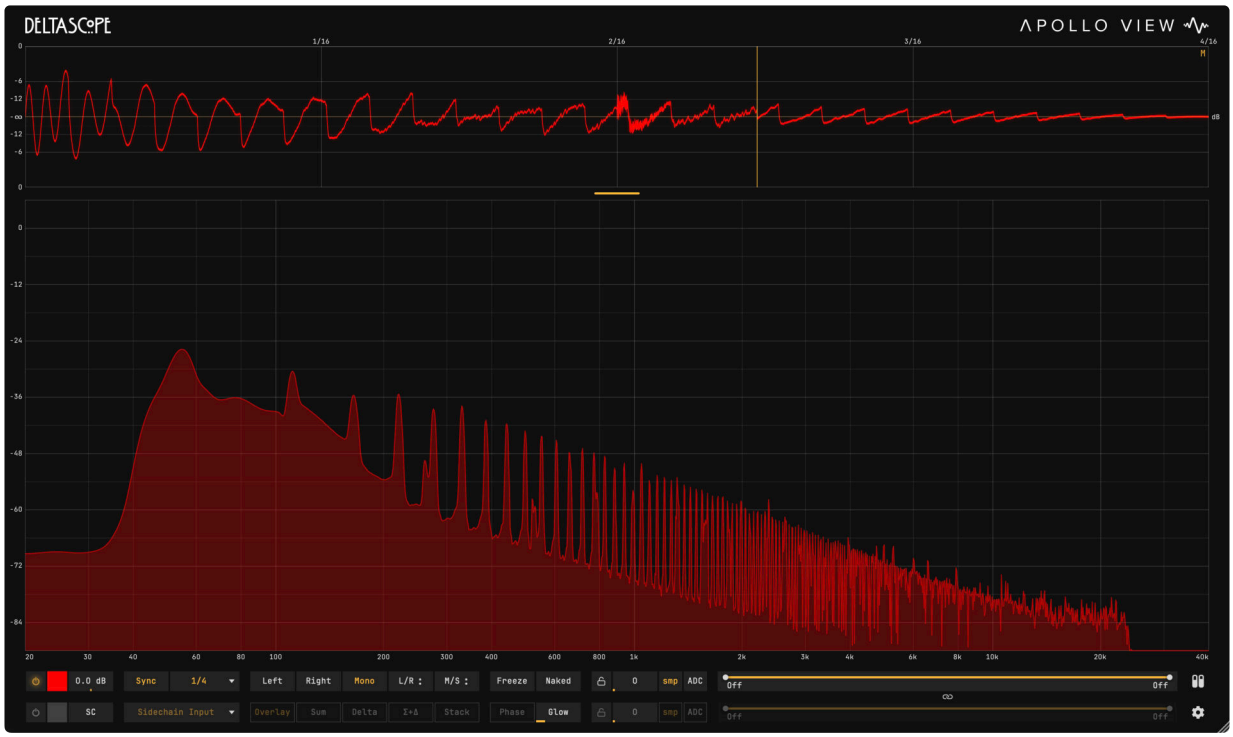
RANGES AND THE NYQUIST LINE

Settings > FFT sets what the pane shows:

- **Frequency range**: `Min`, adjustable 1 to 100 Hz, default 20 Hz; `Max`, adjustable 6 kHz to 96 kHz, default 20 kHz.
- **Level range**: `Floor`, −160 to −30 dB, default −90; `Ceiling`, −24 to +30 dB, default +6.

Widen the level range to watch noise floors; narrow it to compare mix balances.

Raise `Max` above half the project's sample rate and a gold vertical line appears at **Nyquist**, the frequency where the digital spectrum ends. Nothing exists to the right of it. In a 48 kHz project it is at 24 kHz; at 96 kHz, at 48 kHz.



Range extended to 40 kHz in a 48 kHz project: the gold Nyquist line at 24 kHz marks the edge of the spectrum.

A Floor below −92 dB

A deeper `Floor` shows what a float signal chain leaves far below −120 dB, which is where the Distortion readout counts its smallest products.

A loud tone leaves a faint skirt of false level either side of itself, and the usual window lets that skirt hide anything quieter than about −92 dB. So when you set the `Floor` below −92 dB, the display switches to the finer window the Distortion readout measures with, and the depth you see is real right up to each peak. Tone levels are unchanged. Four things follow:

- Peaks are a little wider. At the default `Size` in a 48 kHz project two tones need about 25 Hz between them to separate, where about 15 Hz is enough with the usual window.
- The noise floor reads about a decibel higher.
- Moving the `Floor` across −92 dB restarts the averaging and the peak hold once.
- The bass bands of Multi FFT and the Delta Magnitude view keep the usual window.

RESOLUTION AND MULTI FFT

An FFT trades time against frequency. The longer the stretch of audio it analyses, the finer the frequency detail it can resolve, and the slower it responds. That's the time-frequency trade of any windowed analysis (the uncertainty principle), not a limit of DeltaScope. Separating two tones a semitone apart at 40 Hz (they differ by about 2.4 Hz) needs about a second of signal through the analyser's window. In a 48 kHz project, only the largest `Size` analyses that much audio.

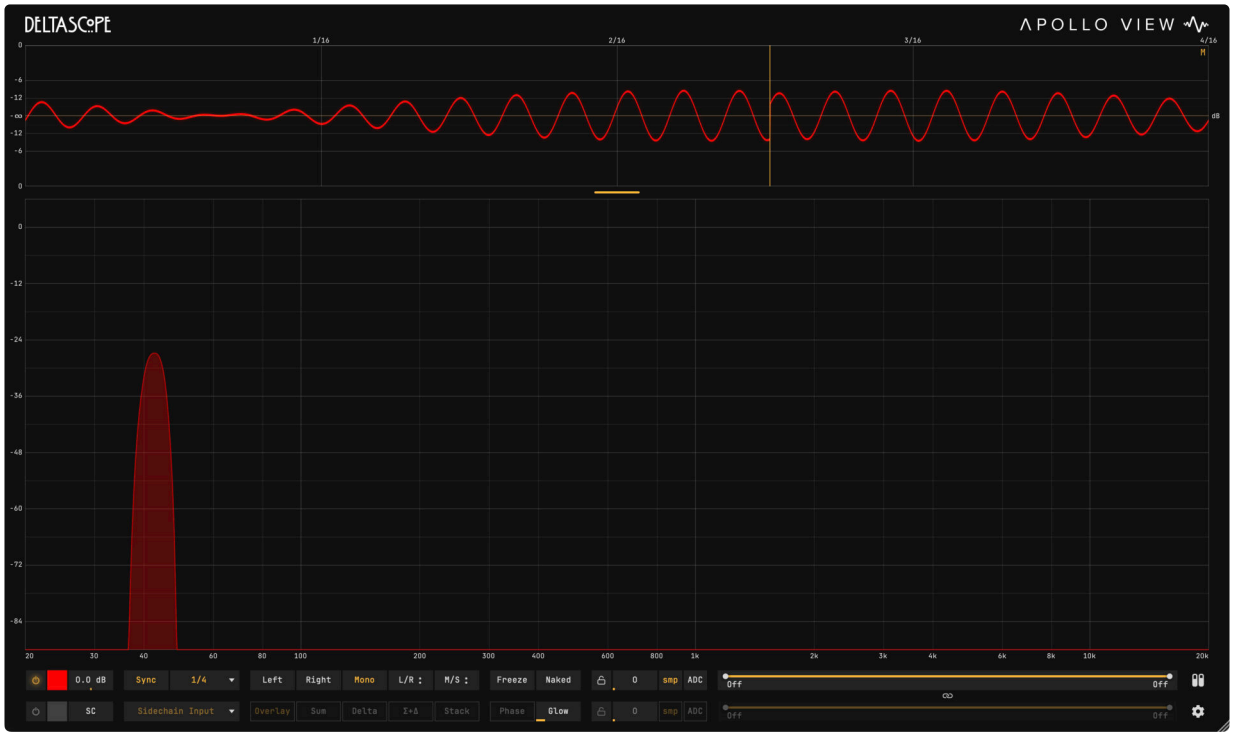
So a single `Size` forces a choice: fast response with poor low-end detail, or sub-bass detail with a slow display everywhere. `Multi FFT` in Settings > FFT removes the choice. With it on, the spectrum is split into up to four frequency bands:

- The top band uses your chosen Size. Everything above the first crossover keeps the speed of your chosen Size.
- Each band below it uses a longer window than the band above, up to three longer windows in all.
- Each lower band starts where the shorter window of the band above no longer has enough resolution, with a half-octave crossfade between them.
- The crossovers move with Size and sample rate.
- At the largest Size there is no window longer than the one you chose, so Multi FFT changes nothing there.
- The extra windows go only three steps longer than your chosen Size, so with the two smallest Sizes the longest window is still too short for the sub-bass.

Two sustained bass notes that merged into one blurred peak at your chosen Size now separate cleanly, and the rest of the spectrum never slows down to make it happen.



Two sub tones one semitone apart (41.2 and 43.65 Hz), Multi FFT off: one blurred lump.



The same two tones with Multi FFT on: two distinct peaks.

The low end responds later while Multi FFT is on. A long window has to see most of a note before it can draw it, so a new bass note appears in the lowest band about half a window after it starts, up to about 0.8 s at the longest window, while the mids show it within a tenth of a second. The bands redraw as often as the rest of the display; the delay is the window, not a fault.

The shape of the window matters as well as its length. Every window spreads a tone into a peak with skirts either side, and shapes differ in how narrow the peak is against how far the skirts reach. The main band uses a Blackman-Harris window, whose skirts stay below the display floor everywhere, so a loud tone never grows a visible base.

The Multi FFT bands use a Blackman window instead: its peak is narrower, so two close bass notes separate when they're about 15 % closer in frequency, and a quieter note beside a louder one gets more room still. Its skirts reach a little further, but at these long windows that's a few hertz beside a loud tone, well below the peak.

In a 48 kHz project at the default `Size` the bands take over at about 250, 125 and 63 Hz, and the longest window separates two equal sustained tones a semitone apart down to about 34 Hz.

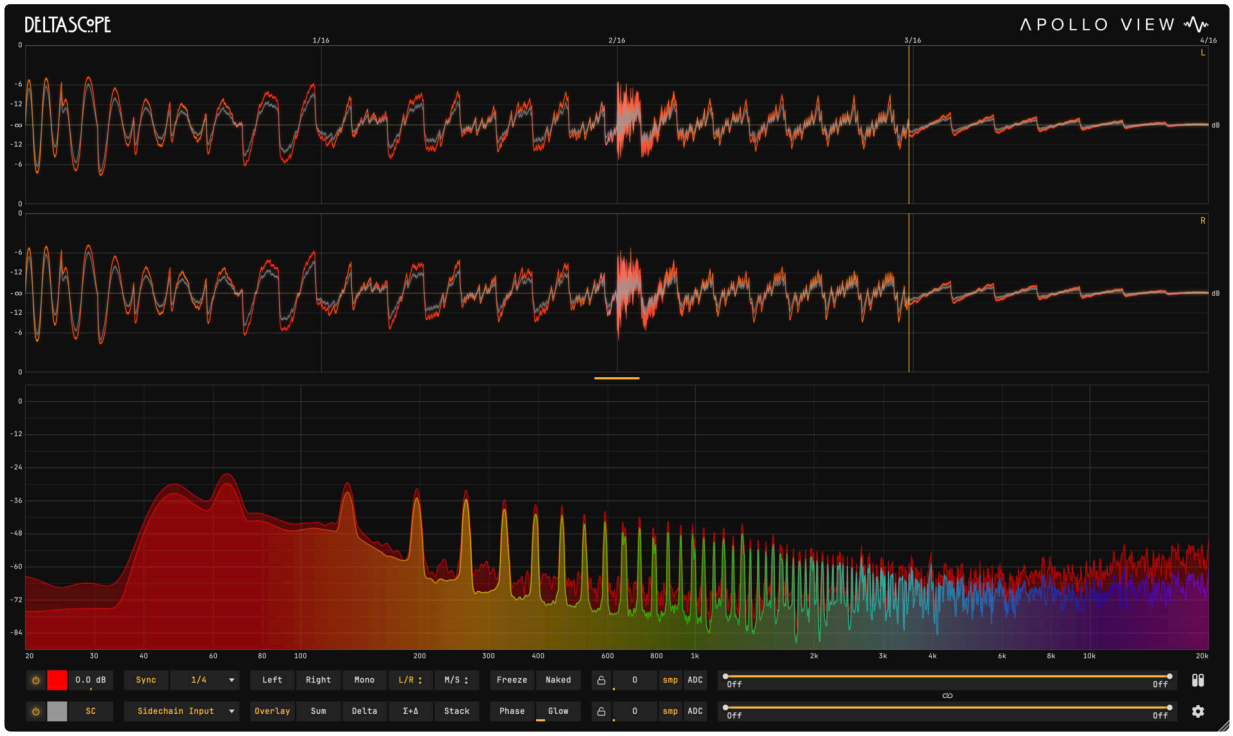
Multi FFT applies to Delta's Magnitude view too. A narrow low-frequency boost or cut is only a few hertz wide, so a short window draws it lower and wider than it is. With Multi FFT on it reads close to its true height.

COLOUR BY FREQUENCY IN THE SPECTRUM

Settings > FFT > Colour by Frequency draws a curve's fill and outline as a run of colour along the axis, the peak-hold line included: red up to 80 Hz, then through orange, green and blue to violet at the top. The region of the spectrum you're looking at then reads by colour as well as

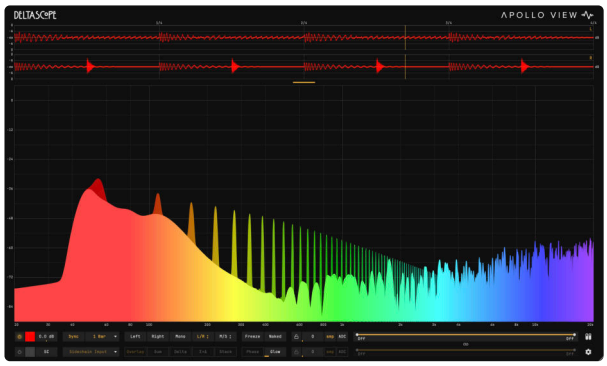
by position. The colours belong to frequencies, not to the pane: zoom the axis and 1 kHz stays green.

- **The switches**: `Main` and `Sidechain` have their own switches. In L/R or M/S the two channels take the same colours, one a little darker and one a little lighter, as they do in flat colour. With a switch off, that curve wears its swatch colour whether or not the scope is coloured.
- **The picker**: it also sets Colour by Frequency. `By frequency` on a channel's tab colours that channel on the analyser and on the scope together (see Colour). Only the picker can colour one channel of a signal and not the other; the two switches here then show their knob in the middle, with a note naming the channel that is by frequency.
- **The sidechain**: in L/R and M/S it is drawn as one curve, which follows the sidechain's `L (M)` tab; in Right its curve follows the `R (S)` tab, as the scope does.

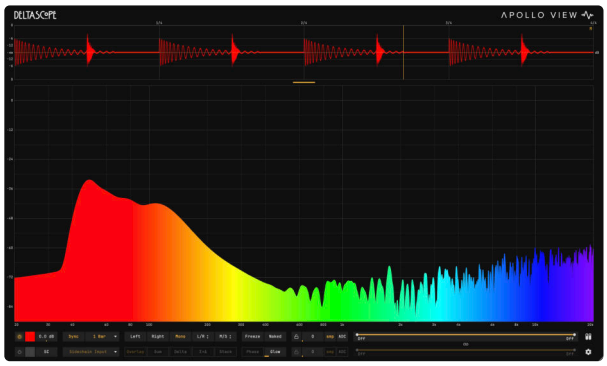


All four switches are independent. Here the scope colours the main signal and leaves the reference flat, while the analyser does the opposite: flat main curve, coloured reference. Neither signal is coloured on both displays, so both swatches show their flat colour.

PRO TIP: In L/R and M/S the lighter curve (right or side) is drawn in front of the darker one (left or mid). A darker see-through fill over a lighter one blends into it as `Fill` rises, so the lighter curve goes in front and both stay readable at any `Fill`. The side channel is usually quieter than the mid, so in Mid/Side it would otherwise sit hidden behind the mid. Switch the channel mode to `Mono` when one curve for both channels is all you need.



L/R at 100 % `Fill`: the lighter right-channel curve in front, the darker left channel showing where it rises above it.



The same audio in Mono: one curve, full-strength colour.

CROSSHAIR MEASUREMENT

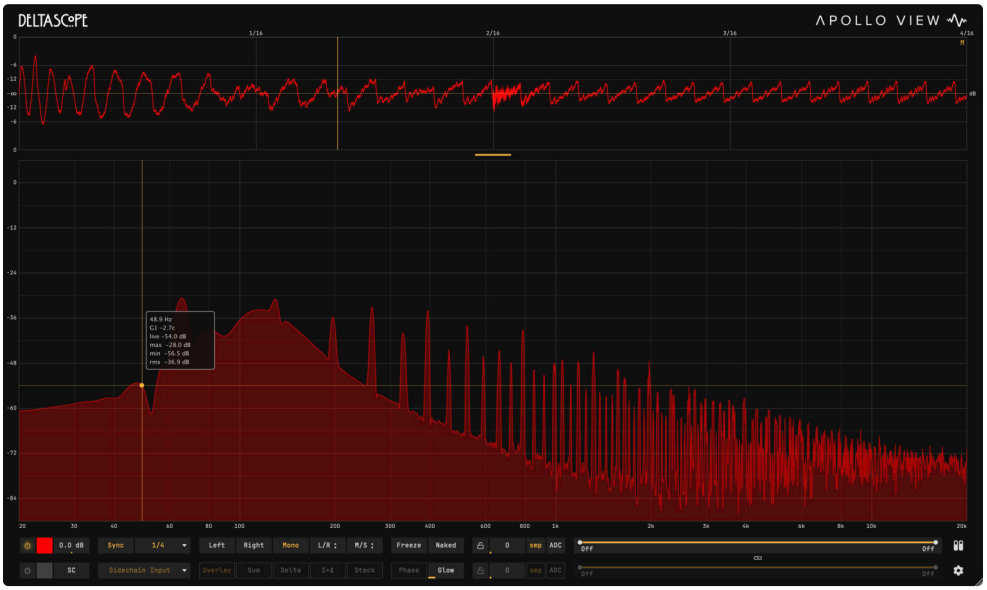
Move the mouse over the spectrum and the crosshair follows with a readout:

- the frequency;
- the nearest note, with cents;
- the level of the curve as drawn (live, or held while frozen);
- **max**, **min** and **rms** at that frequency. The max and min are a true hold over roughly the last 8 seconds; the rms is a running average with a 2.5-second time constant.

The history runs for the whole spectrum even while the crosshair is hidden, so wherever you hover, the answer is already there.

Hold **Alt** to snap the crosshair to the nearest prominent peak (the closest one, not the loudest), so a quiet resonance beside a loud fundamental is reachable. The `Crosshair` toggle in Settings > FFT keeps it always on, or shows it only while Alt is held.

The crosshair gives you a reading like “B3 +12 cents at −18.4 dB” instead of a guess like “somewhere around 250 Hz”.



Alt-hover snapped to the bass fundamental: 48.9 Hz, a hair under G1, with the live level plus the held max and min and the rms at that frequency.

DISTORTION READOUT

Feed DeltaScope a test tone and it tells you how clean the chain is. Switch `Distortion` on (Settings > FFT) and the spectrum shows a box beside the tone's peak with its THD (total harmonic distortion), THD+N (total harmonic distortion plus noise), IMD (intermodulation distortion) and noise floor. Each harmonic it counted gets a label with its level in dB below the fundamental. With no tone present the box says "No tone". The switch replaces the crosshair readout while it is on.



A 1 kHz tone at −5.9 dBFS with the Distortion readout on. The box beside the peak reads the fundamental and its level; THD, with "6 of 9": six of the nine harmonics in range stood clear of the noise and were counted, and each is labelled with its level in dB below the fundamental (-50, −40, −52, −44, −56 and −48 for the 2nd to the 7th); THD+N; IMD, "n/a" because one tone has no pair to intermodulate; and the noise floor.

Set up a test

DeltaScope generates nothing, so the tone comes from your DAW.

1. Put a tone generator on a track. A sine gives THD; two sines give IMD.
2. Run it through the plugin or chain you want to measure.
3. Put DeltaScope after it.
4. Keep the tone at −60 dBFS or louder (with two tones, the louder one) and at 25 Hz or higher.
5. With two tones, keep the quieter within 20 dB of the louder, or it isn't taken as a second tone.
6. Wait for the first reading. It settles in about three seconds, or about twelve for some custom pairs of tones (see What the IMD box reads). The box shows "settling" after the level until then.

The box also says "No tone" when the tone, or the pair, holds less than half of the power in the band. A test tone under music or loud noise reads "No tone" for that reason.

What is measured:

- Freeze measures the frozen moment.
- In L/R both channels are measured and the worse one is shown, with an L or R after the level.
- Left or Right alone measures that channel.
- Mono measures the mono average, (L + R) / 2, and Mid-Side measures Mid.

What the numbers mean

Reading	What it is
THD	The harmonics two to ten, summed, against the fundamental, as a percentage and in dB.
THD+N	Everything except the fundamental, from 20 Hz to just under Nyquist, against the fundamental. It can't read lower than THD.
Floor	The level of the noise between the harmonics, in dBFS, over the same band.
IMD	With two tones, THD and THD+N read "n/a" and IMD takes their place, with the standard's name beside it (see IMD standards).
6 of 9	The harmonics counted, of the harmonics in range.

The fine print

- **In range**: only harmonics that fit below Nyquist are in range, and only those standing clear of the noise are counted. The box shows "n of m", counted of in range. A 10 kHz tone at 48 kHz has just one harmonic in range, and the box says so.
- **Counting**: a harmonic counts only when it sits 6 dB above the noise floor and at −140 dB or higher against the fundamental. The noise floor is one estimate for the whole band, the average of the bins that are not part of a peak, taken over the width of one peak. Every harmonic is compared against that same figure, not against the noise beside it.
- **Digital sines**: a plain digital sine carries rounding harmonics near −150 dB, so none is counted and THD reads a bound, such as "< −132.4 dB 0 of 9", rather than inventing a figure.
- **Percentages**: three significant figures, to at most four decimals. Anything under 0.0001 % reads "< 0.0001 %", with the dB figure beside it carrying the detail. On the THD+N row, a bare sine from a DAW reads "< 0.0001 % −121.9 dB".

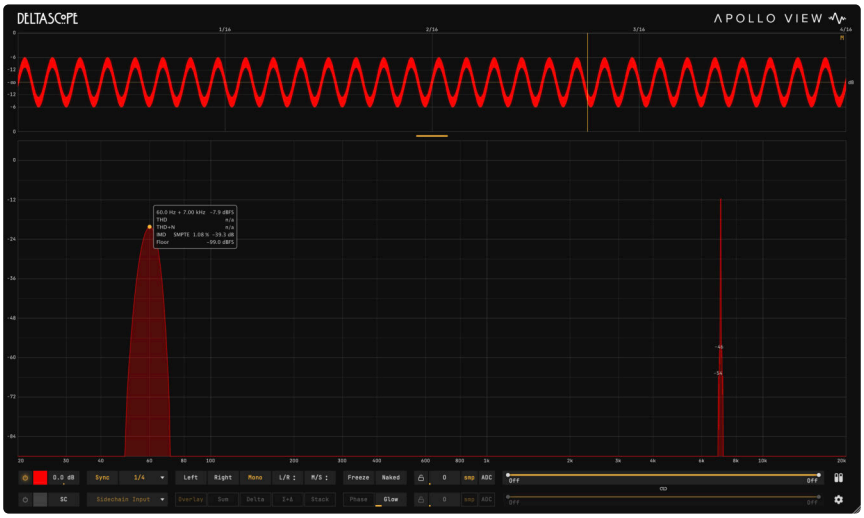
IMD standards

Two tones give IMD. Call the lower tone f1 and the higher f2. DeltaScope recognises three standard pairs and reads any other pair as “custom”. Each row says what the two tones must be for the box to name that standard.

Standard	The two tones	Level	The figure	Usual test
SMPTE	Low tone under 150 Hz. High tone at least eight times its frequency.	Low tone 6 to 18 dB louder.	The sidebands around the high tone (the products at the high tone plus and minus whole multiples of the low tone), against the high tone.	60 Hz and 7 kHz, low tone 12 dB up.
DIN	Low tone at 150 Hz or above. High tone at least eight times its frequency.	Low tone 6 to 18 dB louder.	The sidebands around the high tone, against the high tone.	250 Hz and 8 kHz, low tone 12 dB up.
CCIF	Both tones at 1 kHz or above, at least 25 Hz apart, within 1.25 to 1 in frequency.	Within 3 dB of each other.	The difference products (f2-f1, 2f1-f2 and 2f2-f1), against the sum of the two tone amplitudes.	19 kHz and 20 kHz.
custom	Any other pair.	The quieter tone within 20 dB of the louder.	Every sum and difference product from the second to the ninth order, summed as RMS against the sum of the two tone amplitudes.	An example: a fifth, 100 Hz and 150 Hz, is mostly the 50 Hz difference tone.

The box reads the name, the percentage and the dB figure, as in “SMPTE 1.08 % −39.3 dB”. Other analysers divide by other things, so say which formula you used when you compare. A product counts by the same rule as a harmonic: 6 dB above the noise floor, and at −140 dB or higher against its reference.

Keep the high tone off exactly 8, 9 or 10 times the low tone (250 Hz and 2 kHz, for instance): DeltaScope takes a tone within about 3 Hz of those as a harmonic of the low tone, and the box reads THD instead of IMD.



A 60 Hz and 7 kHz pair, the low tone 12 dB above the high one, with the Distortion readout on. The box names both tones; THD and THD+N read “n/a”; IMD reads “SMPTE 1.08 % −39.3 dB”; and the noise floor.

In the figure the sidebands sit 60 and 120 Hz either side of the 7 kHz tone, and are labelled −46 and −54 against that tone. The box reads 1.08 %, which is higher than either label suggests, because the two sidebands of each pair add, and the orders then add as RMS. At the default +3 dB/oct slope the 60 Hz peak is drawn lower and the 7 kHz peak higher than their levels.

What the IMD box reads

The row reads	What it means
`custom  1.77 %  -35.0 dB`	The figure is nearly complete: neither the products under the noise floor nor those left out beside a harmonic could add more than half a dB to it.
`custom  at least 1.77 %  -35.0 dB`	The figure counts only the products it could measure. This happens where no window length separates a product from a harmonic, as with a fifth, and where products under the noise floor could add more than half a dB. In L/R “at least” also shows whenever either channel’s figure is a lower bound or could not be measured.
`custom  below floor`	Products are in range, but none stands clear of the noise.
`n/a`	Every product lands on a harmonic, as with an octave whose upper tone is the louder (100 Hz and 200 Hz, the 200 Hz louder), or none could be counted but the places left out hold something. If the lower tone of that octave is the louder, the 200 Hz is read as the second harmonic of one tone instead, and with the two at equal level the box can read “No tone”. A standard pair reads “n/a” too when none of its products is in range.

A custom figure leaves out any product that lands on a tone or on a harmonic of either, because it can't be told from the harmonic. A custom figure belongs to the pair you played, so don't compare it with a datasheet. For musicians it is the figure for two notes of a chord through a saturator: it tells you how much the notes intermodulate.

Some custom pairs take longer to read. When a product sits too close to a harmonic to be told apart, but a window four times as long separates them, DeltaScope measures over that longer stretch. 60 Hz and 7 kHz at equal level is one such pair. It takes about twelve seconds to settle instead of three. The box shows the longer measurement as it builds, marked "settling", until it is complete.

Point at a harmonic or a product
Move the mouse over the spectrum. When it is within 12 pixels of a counted peak, the box and a gold dot jump to that peak, and the box's first line names it with its level. A level in dBc is in dB relative to the tone the peak belongs to, here the fundamental.

First line	What it is
`3rd  3.00 kHz | -75.0 dBFS  -63.0 dBc`	A harmonic.
`f2-f1  50.0 Hz | -35.0 dBFS  -30.0 dB`	An intermodulation product. Its dB is relative to the same reference as the IMD figure.
`f2  150.0 Hz | -12.1 dBFS`	The quieter tone, named by its place in the pair: f2 when it is the higher, f1 when it is the lower.
`5th of f1  865.0 Hz | -25.0 dBFS  -22.0 dBc`	With two tones, a harmonic of one of them, with the dBc against that tone's own level. It is not part of the IMD figure and has no label on the display; it only answers the pointer.
`near 11th of f1`, `near 3f1+2f2`, `near f2-f1`	A peak the readout does not count (an order above the ninth, a harmonic above the tenth, an alias, hum): the best guess at its name, then the frequency and the level in dBFS taken live from the curve, with no dBc.
`peak`	The same, when no combination of the two tones fits.

How the box behaves

- The four summary rows stay. The peak's own small label is not drawn while the box names it.
- The box stays on the last peak you pointed at until the pointer reaches another one, so you can move off the curve without losing the reading.
- Leave the display and the box goes back to the fundamental. Point at the fundamental (with two tones, the louder one) and it holds there.

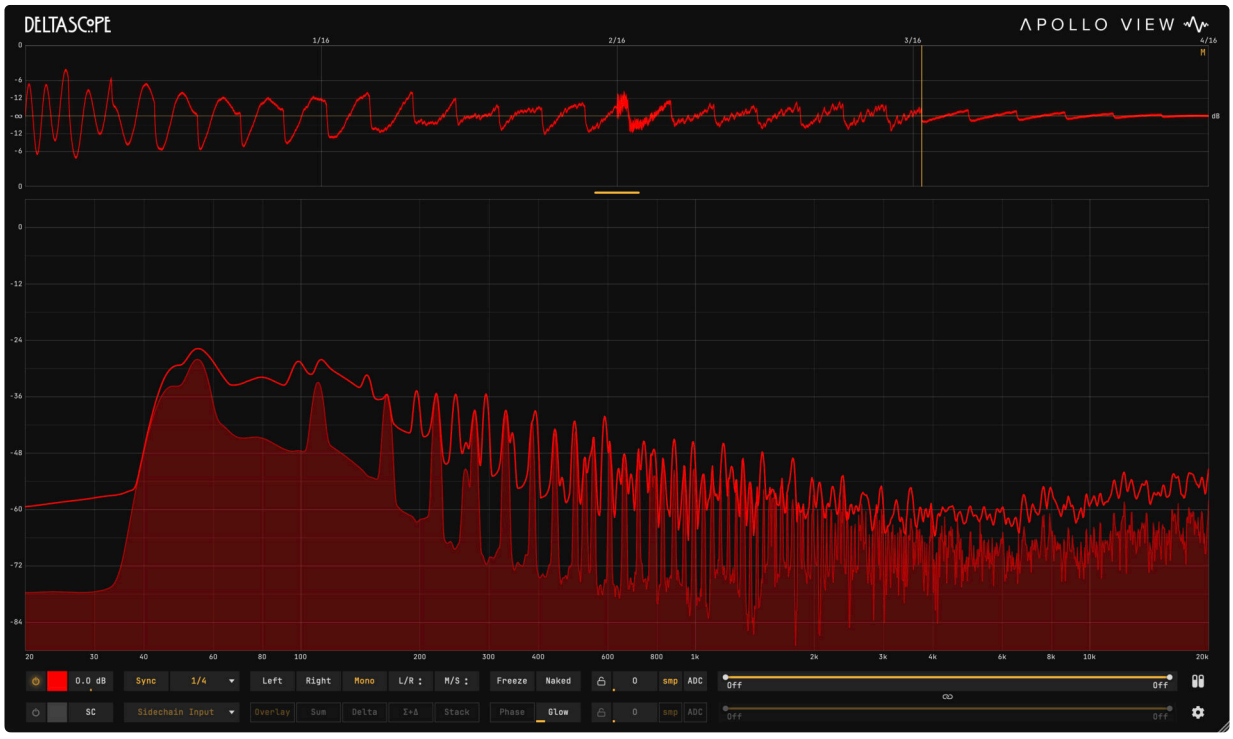
- If a new reading no longer has that peak, the box goes back to the fundamental until you point at a peak again.
- The gold dot and the peak labels sit on the curve as it is drawn right now, so they follow a moving peak. The figures in the box are the averaged measurement of the last three seconds or so (about twelve for a pair on the longer measurement).
- A peak the readout does not count is picked up only if it stands at least 6 dB above the lowest point within 12 pixels, so the noise floor is never picked up. A counted peak within reach always wins.
- It works whatever the `Crosshair` switch says. With “No tone” the mouse does nothing.

What it can’t do

- A harmonic reads to a tenth of a dB only when it sits about 24 dB above the noise floor. Closer to the noise the reading wanders, by up to a dB at the 6 dB edge.
- A tone that moves, from a pitch shifter or a modulated chain, won’t hold still long enough to measure.

PEAK HOLD

Transients are too brief to read from a moving curve. A snare hit is over before you can read it, and a resonance that appears once per bar never shows on an average. The peak-hold line is drawn above the live curve and keeps each frequency’s highest recent level: it holds for the `Hold` time, then falls at the `Decay` rate. The gap between the held line and the live curve is how far your peaks are above the average. Set `Hold` to its top step (or type “inf”) and it never falls: a whole-song maximum trace.



Peak hold on: the filled curve is live; the thin line above it is each frequency’s held maximum.

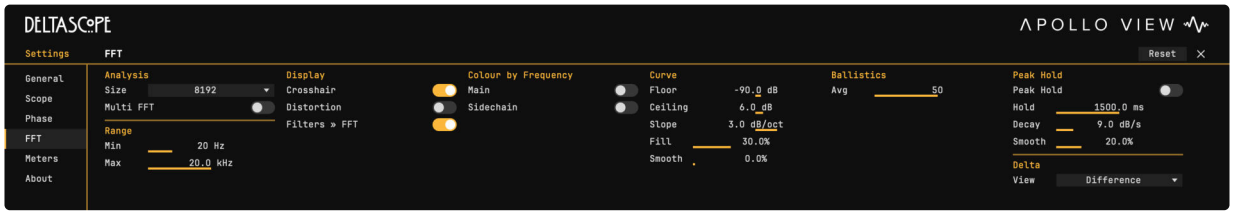
Any FFT display, DeltaScope’s included, treats tones and noise differently:

- **Tones**: a tone shows its true level at any `Size`. With `Slope` at 0, a −24 dB sine reads within a decibel of −24 dB whether it is at 50 Hz or 15 kHz; the last fraction of a dB

depends on where the tone falls between analysis bins (the FFT's frequency steps). At the default +3 dB/oct the display is deliberately tilted around a pivot at 1 kHz, so the same tone reads higher at 15 kHz and lower at 50 Hz. Set Slope to 0 whenever you want absolute levels.

- **Noise**: broadband noise moves with FFT size. Doubling `Size` halves each analysis band's width, so each band collects half the noise power: the noise floor drops about 3 dB per doubling while tones keep the same level. Every analyser does this; a noise level only means something together with the bandwidth it was measured in. The same physics means the noise floor steps down gently through Multi FFT's bands while tones cross every band boundary at the same level, within about a decibel.

FFT SETTINGS (SETTINGS > FFT)



Settings, FFT page.

Section	Control	What it does
Analysis	`Size`	FFT length, 2048 to 65536. Bigger means finer frequency detail and slower response. Default 8192.
	`Multi FFT`	Multiresolution analysis: the low end uses progressively longer windows for sub-bass detail while the rest of the spectrum keeps the speed of your chosen Size (see Resolution and Multi FFT). The low end itself responds more slowly while on. Default off.
Range	`Min` / `Max`	Displayed frequency range. Min 1 to 100 Hz (default 20), Max 6 to 96 kHz (default 20 kHz). Raise Max above Nyquist and the gold line appears.
Display	`Crosshair`	Always-on crosshair readout; off means hold Alt to show it. Default on.
	`Distortion`	Measure a test tone's THD, THD+N, IMD and noise floor, in a box beside the fundamental (see Distortion readout). Replaces the crosshair readout while on. Default off.
	`Filters » FFT`	Band filters also apply to the spectrum (see Band filters). Default on.

Section	Control	What it does
Colour by Frequency	`Main`	Draw the main curve (both channels in L/R or M/S) as a run of colour along the axis, red at the bass through green to violet at the top (see Colour by frequency in the spectrum). If the picker has made only one channel by frequency, the knob sits in the middle with a note, `L (M) only` or `R (S) only`. Click it and both channels turn on.
	`Sidechain`	The same for the sidechain curve, and for the delta curve drawn beside the sum in Σ+Δ. It shows the middle state the same way. In L/R and M/S the sidechain is one curve and wears its first channel's choice; in Right it wears the second's.
Curve	`Floor` / `Ceiling`	Level range in dB. Floor −160 to −30 (default −90), Ceiling −24 to +30 (default +6).
	`Slope`	dB/octave tilt, −9 to +9 in steps of 0.1 (default +3). Music gets quieter by roughly 3 dB per octave as frequency rises, so at the +3 default a balanced mix reads flat and imbalance reads as slope; 0 is measurement-flat.
	`Fill`	Opacity of the shaded area under the curve, 0 to 100 %. Default 30 %.
	`Smooth`	Fractional-octave smoothing of the live curve: at 0 you see every analysis bin, wobble included; raise it and the curve smooths into a tonal-balance read, at the cost of narrow peaks and notches. Default 0.
Ballistics	`Avg`	How quickly the curve follows the music: low for a fast curve that shows every hit, high for a steady average you can compare against. Range 0 to 100, default 50. DeltaScope compensates for the FFT length, so changing `Size` changes the response speed much less than it otherwise would; bigger sizes are still a little slower.
Peak Hold	`Peak Hold`	Enable the peak-hold line (see Peak hold). Default on.
	`Hold` / `Decay`	Hold time before decay, 0 to 10 s (the top step is infinite; default 1.5 s), and decay rate, 1 to 60 dB/s (default 9).
	`Smooth`	Fractional-octave smoothing of the peak-hold line, across frequency, so it reads as a smooth outline rather than a jagged, spiky line. It's never less than the curve's own `Smooth`, and never fully off. Default 20 %.
Delta	`View`	Difference or Magnitude (see Delta and Hear the Delta). Set it at any time; it takes effect in the Delta comparison mode.
	`Reset`	This page back to its defaults.

SIDECHAIN

The bottom control row belongs to the reference signal. `SC` switches it on.



The sidechain row: power, colour, SC, the source picker, the comparison modes, Phase and Glow, and the sidechain's own alignment cluster.

The row has two switches because they do two different jobs. `SC` turns the reference signal on or off. The SC power button only hides the reference's picture, meaning its trace and its spectrum curve; the reference stays connected, so you can clear the picture for a moment and bring it back with one click. Nothing to re-route.

Until `SC` is on, the rest of the row is greyed: the power button, the colour swatch, the five comparison modes, `Phase`, the source picker, the sidechain band filter and its alignment cluster. Only `Glow` stays active, since it needs no reference. Even dimmed, the comparison-mode buttons still show which mode will come back when you switch `SC` on again.



The sidechain row with `SC` off: only `Glow` is active.

CHOOSING THE SOURCE

The source picker, active once `SC` is on, offers two kinds of reference:

- `Sidechain Input`: the plugin's sidechain bus, routed from your DAW like any sidechain compressor's key input.
- **Another DeltaScope instance**: every DeltaScope of the same plugin format in the project appears in the menu by track name (as "Instance n" where the host reports no names). Pick one and DeltaScope receives its signal directly.
  - Once the transport has run, the two signals are lined up by host position, so the DAW's own delay compensation and processing order are handled for you.
  - Whether the latency of plugins inserted between the two instances needs typing in depends on your DAW (see Alignment, Latency between two linked instances); where it does, the offset and ADC controls are for that, and on a linked pair ADC finds it from the first sound after play, or from the music as it plays (see ADC on a linked pair).

> **NOTE:** Two instances link only when they run in the same process, and a host that runs each plugin in its own separate process keeps instances from finding each other. In Bitwig Studio the plugin hosting mode decides this. Its default, `Together`, keeps DeltaScope's instances together, so they link; `Individually` hosts every instance in its own process, so a pair can't link.

> **NOTE:** Instance linking is the quickest way to compare two points in a chain: insert a DeltaScope before the processing and one after, then link the second to the first.

NOTE: The instance you link to has to be running. Bypass or disable it in your DAW and the instance that listens to it has nothing to subtract: its Delta shows and plays the whole main signal until the other instance runs again.

NOTE: The two sources differ in one thing you can hear.

- With `Sidechain Input` the host hands DeltaScope both signals together, so they are compared as they arrive, with nothing added to line them up: route the reference into it as you would a compressor's key.
- Linking another instance needs no routing. The price of that convenience is that its Sum and Delta (see Comparison modes) can be heard and drawn behind the live track, because DeltaScope reads the other instance a little behind the live signal: at least 20 ms, in whole audio buffers, and further behind when the host is busy. An instance that sits earlier on the same track, when the host never runs it late, adds no extra delay (see Hear the Delta).

Pro Tools: the key input is mono (a Pro Tools convention that applies to every AAX plugin) and DeltaScope uses that one channel for both lanes. Overlay and Stack work as anywhere else. Sum, Delta and Σ+Δ compare each main channel against the same mono key, so on stereo material the delta includes the stereo difference itself. For a true stereo delta, link another DeltaScope instead: Link carries stereo and takes precedence over the key input. Keep the key input for mono references.



The source picker: Sidechain Input, then every other DeltaScope of the same format in the project by track name. Here there is one, on the track called Reference. Pick one and its audio is sent directly to this instance: compare any two points in the project, even across tracks the DAW can't sidechain.

Links are made by track name: reopen the project and every instance reconnects by name.

- Ableton Live's automatic track numbering, a number and a space before the name, is stripped, so renumbered tracks still match.
- If renumbering leaves a name matching more than one track, DeltaScope leaves the link unresolved rather than guessing.
- A host that allows two tracks with literally the same name is different: there the first match is used, so keep names distinct.
- Delete a source and the link keeps the name, so an undo reconnects it.

No routing to set up, and nothing to set up again when you reopen the project.



Your mix (red) against a commercial reference (grey), overlaid live. No sidechain routing needed: the two instances are linked by track name. The gap between the two curves shows how much louder the reference is than the mix at each frequency; it is widest in the top octaves.

COMPARISON MODES

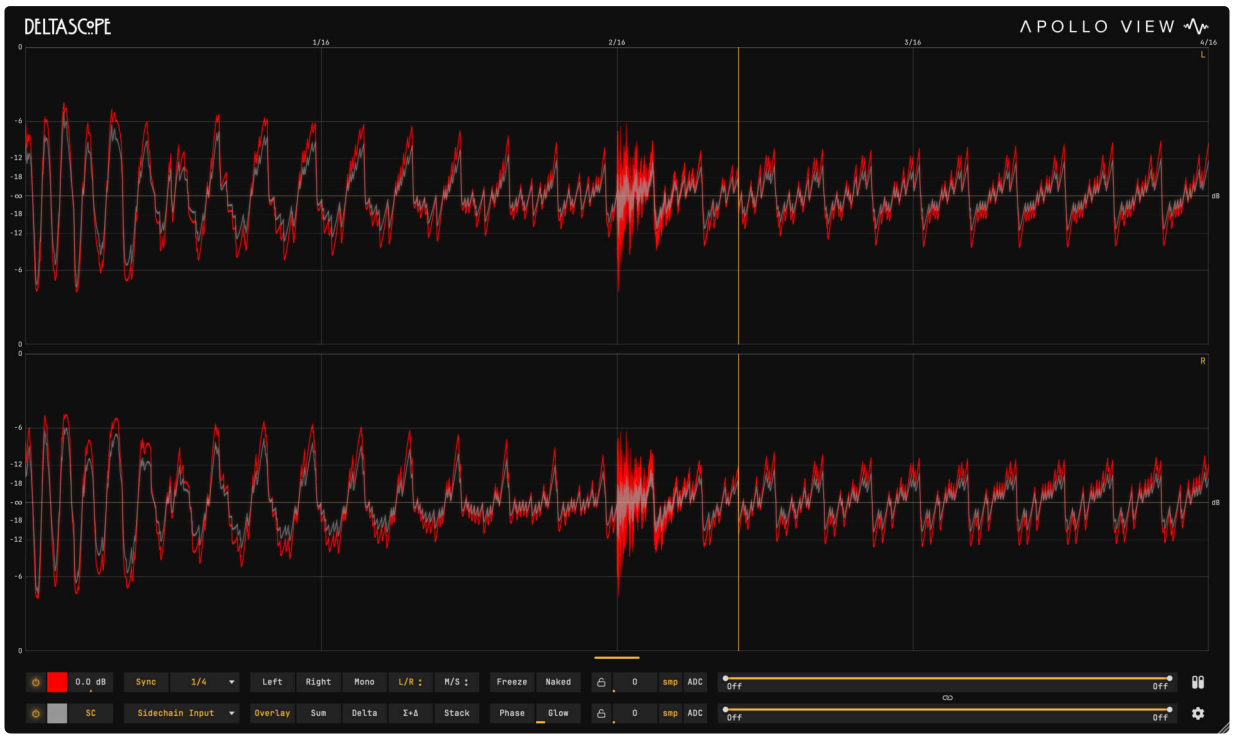
Mode	Shows
`Overlay`	The reference drawn over the main waveform in its own colour: where the traces separate is what's different.
`Sum`	Main plus reference: the two added together. Watch kick and bass combine, or a dry track with its parallel return, and see where the sum gains or loses level.
`Delta`	The difference, main minus reference. Identical signals draw a flat line.
`Σ+Δ`	Sum (Σ) and Delta (Δ) in stacked lanes: the two together and the difference between them, on screen at once.
`Stack`	Main and reference in separate stacked lanes. Use it instead of Overlay when the overlapping traces are too dense to read.

> **NOTE:** `Σ+Δ` and `Stack` use both display lanes for the comparison, so an L/R or M/S channel mode switches to Mono, and the `L/R` and `M/S` buttons are unavailable while either is active (Left, Right and Mono stay as chosen). `Delta` on its own keeps whichever channel mode you've chosen and subtracts lane by lane (see Per-lane delta in Delta and Hear the Delta).

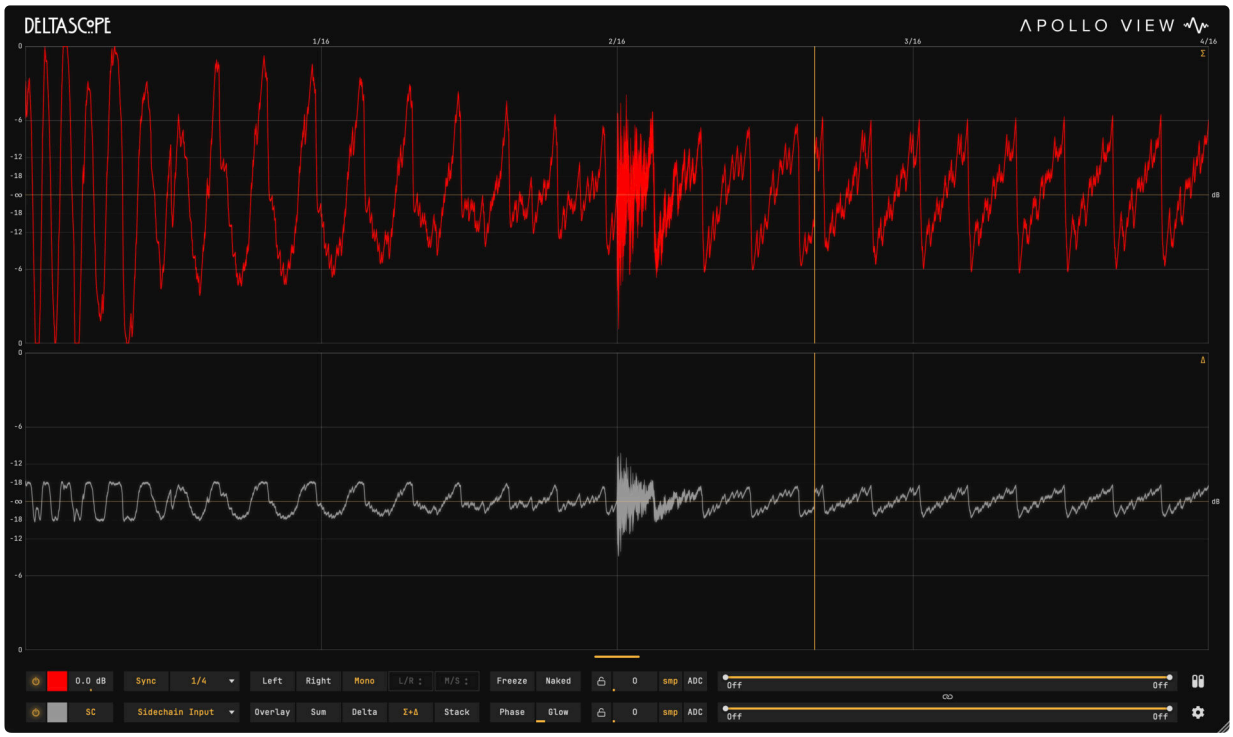
Below 1230 pixels wide, the five buttons fold into one dropdown, at the same width as the channel buttons in the top row, so the two rows stay in line. The band sliders keep their width. While a comparison mode is on, the top row's dropdown lists plain `L/R` and `M/S`, greyed under `Σ+Δ` and `Stack`, as the buttons are.



The same dropdown with `Stack` on: it lists plain `L/R` and `M/S`, greyed, and the view has switched to `Mono`.



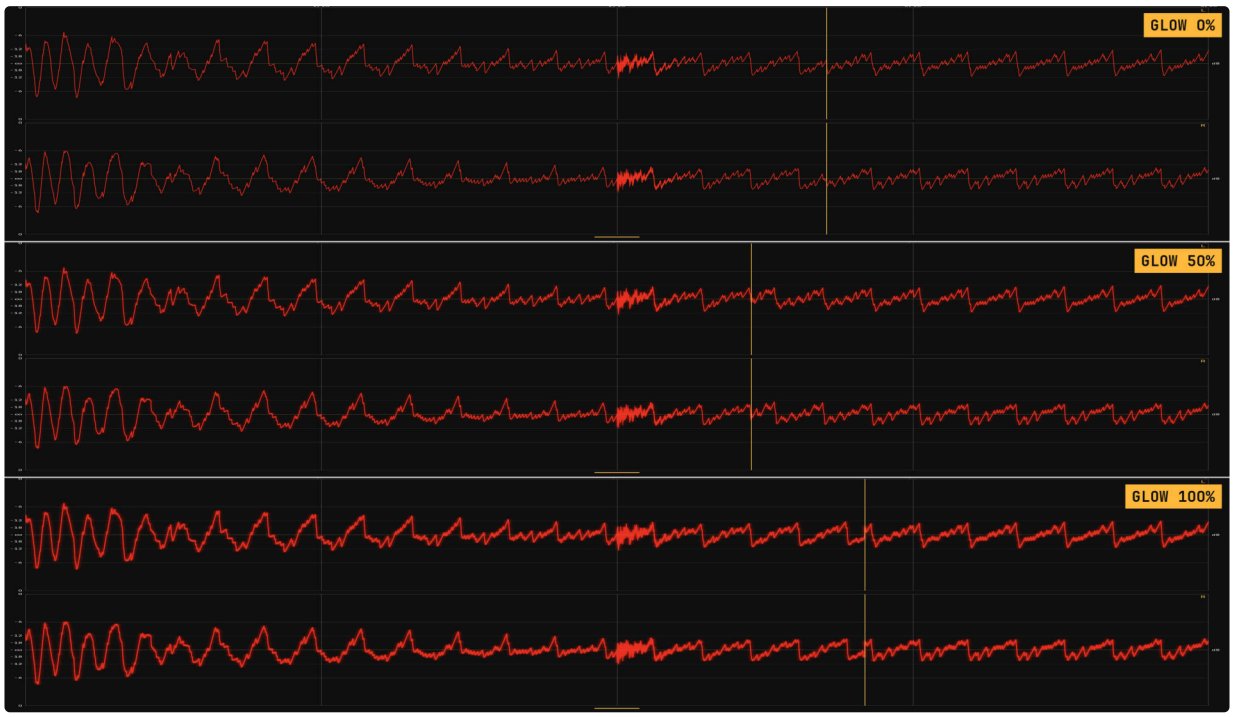
Overlay: processed (red) over dry (grey). Where the traces separate is what the processing is doing.



Σ+Δ: the Σ lane above carries the sum, the Δ lane below the delta, both built from the same two lined-up signals.

GLOW

The `Glow` slider adds a phosphor-style bloom to the waveforms. Like the phosphor glow of a CRT oscilloscope, it's a soft blur of the trace, brightest where the trace is densest on the display. It's a visual effect only; it doesn't measure anything. Alt-click or double-click resets to 0.



The same beat at Glow 0%, 50% and 100%.

DELTA AND HEAR THE DELTA

DeltaScope subtracts the two aligned signals once, as the audio passes through, and keeps the result as a separate signal.

> **NOTE:** The trace, the spectrum's Difference view, the measurement readouts and the `Hear the Delta` output all work from that one difference signal. What you see is what you hear, and an exported image of a delta lane shows the same signal that left the plugin's output. Two things change the picture without changing the signal: the display gain, and the band filter. With a band active you see a band-limited delta and hear the full-band one.

READING A DELTA

Subtract the original from the processed signal and only the change is left. What that change looks like depends on the processor:

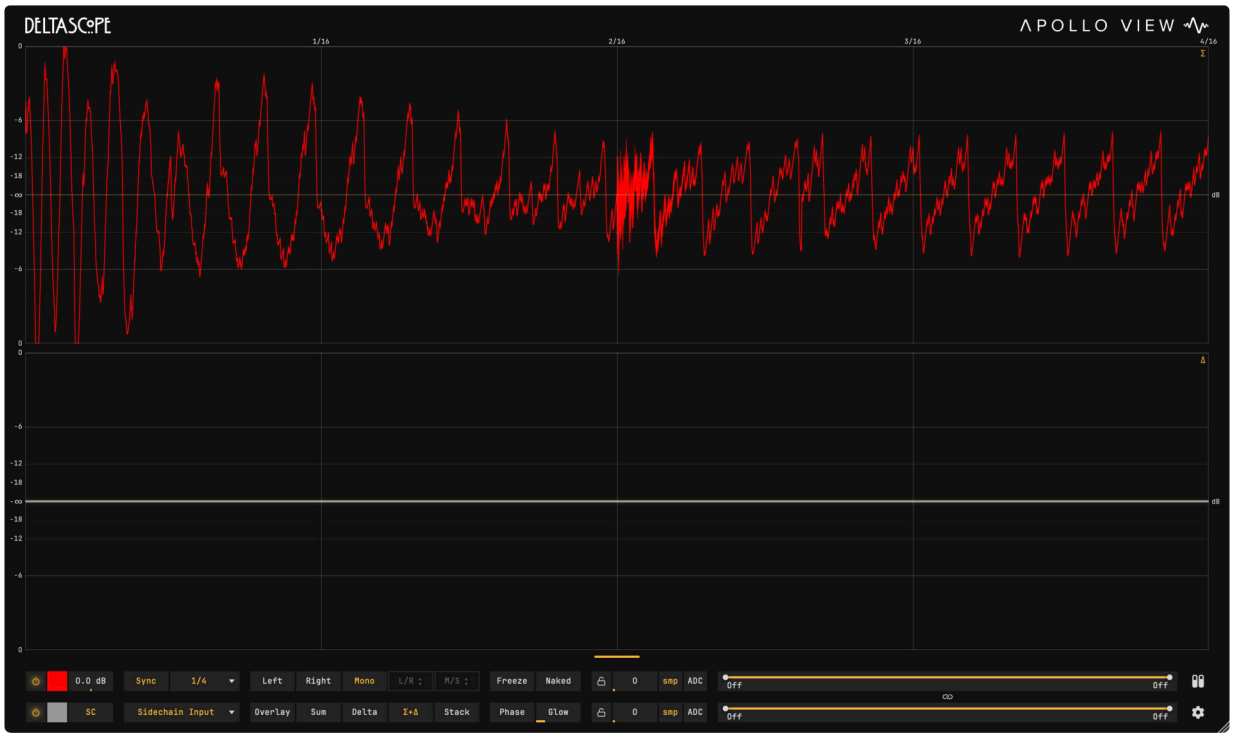
- **An EQ**: the delta shows what the curve is really doing, level and timing together. Most EQs are "minimum-phase": they shift the frequencies near each band slightly in time as well as changing their level, and that time shift appears in the delta too.
- **A compressor**: the delta is what the compressor is changing, drawn in time. With make-up gain off, it's the part of the signal the compressor is taking away. Its envelope rises and falls with the gain reduction, and it is larger where the signal itself is louder. You see how much of each transient passes before the gain reduction starts, and how far the gain recovers between hits. That's the compressor's attack and release, as a waveform.
- **Which frequencies a processor changes most**: open the spectrum in Delta mode and the peaks are the frequencies where the two signals differ most, in level or in timing. The height of each peak also depends on how loud your material is at that frequency, so a small change on a loud bass can show a taller peak than a larger change on quiet high frequencies. A peak means *changed*, not necessarily removed: an exciter adds, a compressor mostly takes away. Either way, the delta shows you where the change carries the most energy. The Magnitude view removes that dependence on the material's level.

THE PERFECT NULL

The null test is the classic way to check whether two signals are identical: subtract one from the other and listen to what's left. Traditionally you flip the polarity of one copy, play both together and listen for silence. DeltaScope does the same subtraction live and keeps the result on screen while you play.

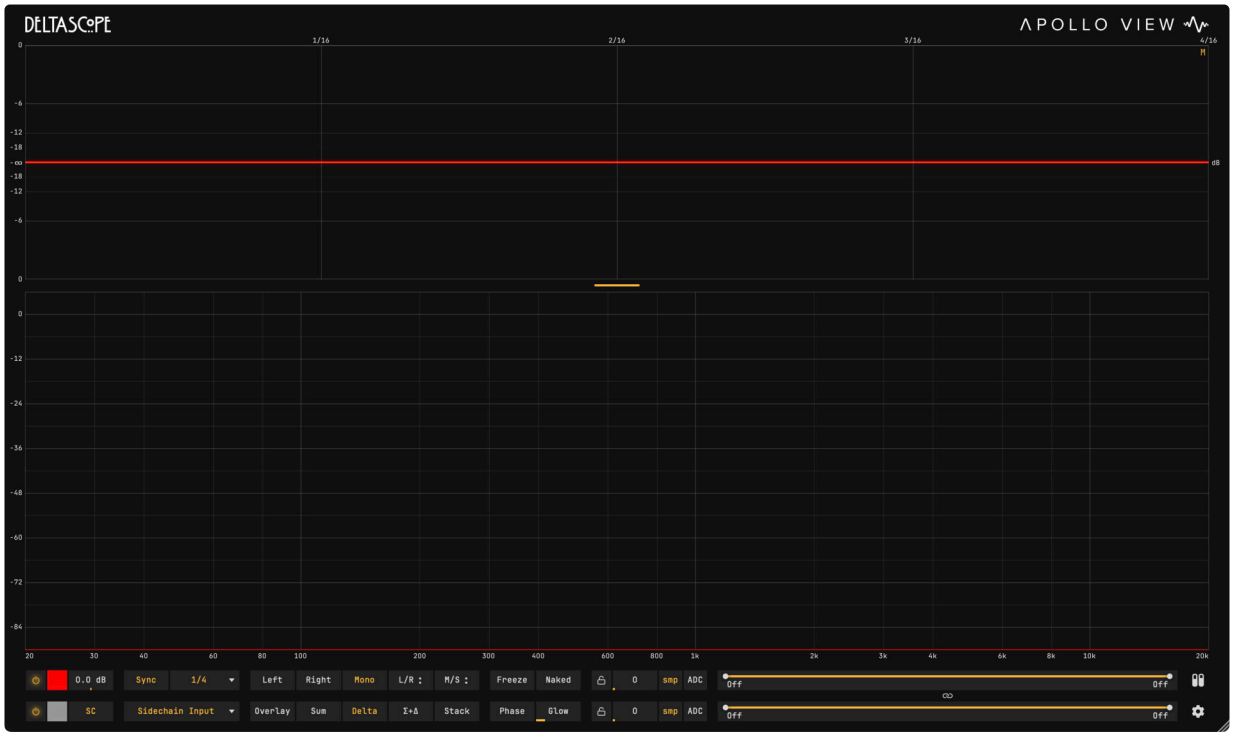
1. Route the second signal in, from the sidechain input or another instance picked by track name.
2. Click `Σ+Δ`.
3. Read the two lanes.

The Σ (sum) lane carries the two added together; the Δ (delta) lane is what differs. Identical signals cancel completely and draw a flat line, and the display keeps updating on real programme material while you listen.



The null test, live: the Σ lane carries the two signals added together, the Δ lane the difference between them. Here the two are identical, so the sum is the mix doubled and the delta a flat line.

Click `Delta` to give the residual (what's left after the subtraction) the whole display, and open the spectrum's Difference view to check it across the whole range at once. If the null is close but a small, unsteady residue remains instead of a flat line, the two signals are most likely slightly out of time rather than different. See Alignment.



The same null through the analyser's Difference view: the spectrum of what's left is at the bottom of the level range at every frequency, nothing above it anywhere from 20 Hz to 20 kHz.

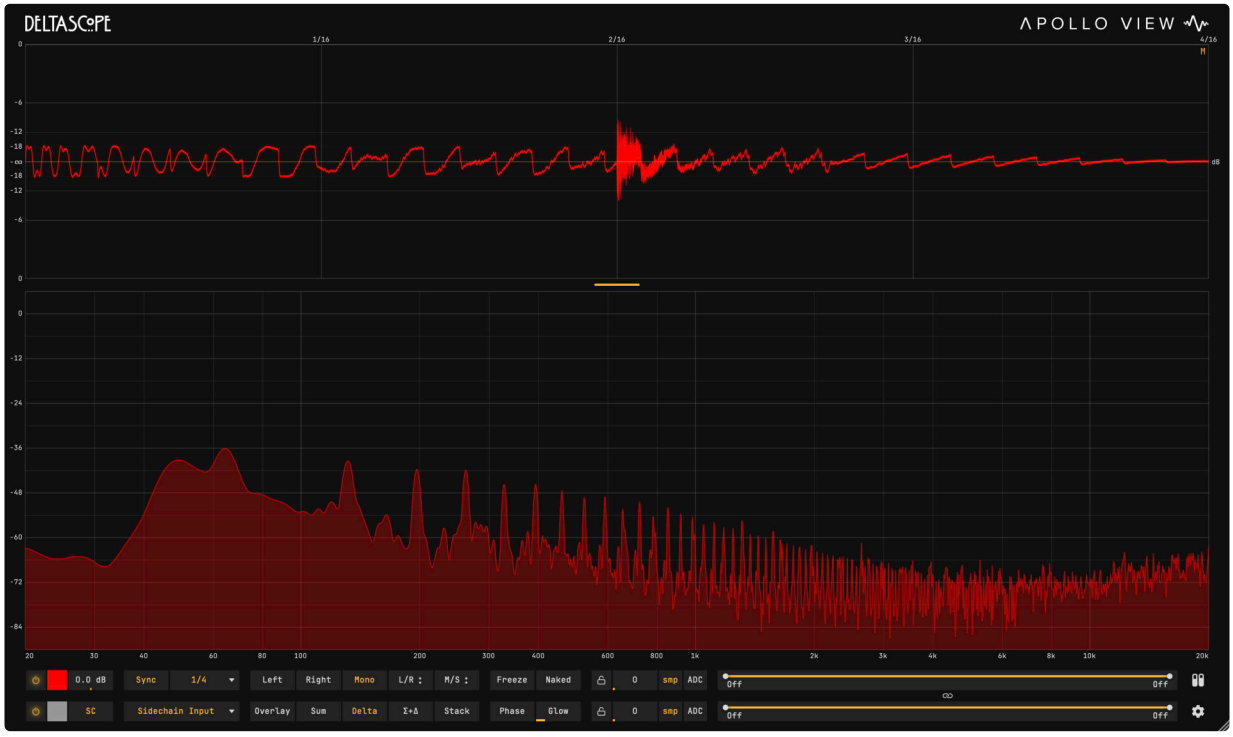
PER-LANE DELTA

Delta respects the channel mode. In Left, Right or Mono there's one delta; in L/R you get an L-delta and an R-delta; in Mid-Side, a mid-delta and a side-delta, so you can see whether a processor changes your side channel differently from your mid. The spectrum's Delta views draw the same lanes: on `Auto`, a darker shade of the main colour for L/Mid and a lighter one for R/Side.

THE DELTA IN THE SPECTRUM: DIFFERENCE AND MAGNITUDE

In the Delta comparison mode the spectrum can show two different things. Choose with `View` in Settings > FFT. You can set it at any time, whatever the comparison mode; it takes effect once `Delta` is the comparison mode:

- **Difference**: the spectrum of the delta itself, the same signal the scope draws: it shows how much is changing and at which frequencies.
- **Magnitude**: how much louder or quieter the main signal is than the reference at each frequency, in dB. That's the frequency response of whatever processing is between them: how much it boosts or cuts. It's a ratio, so it reads the same whether the music is loud or quiet.



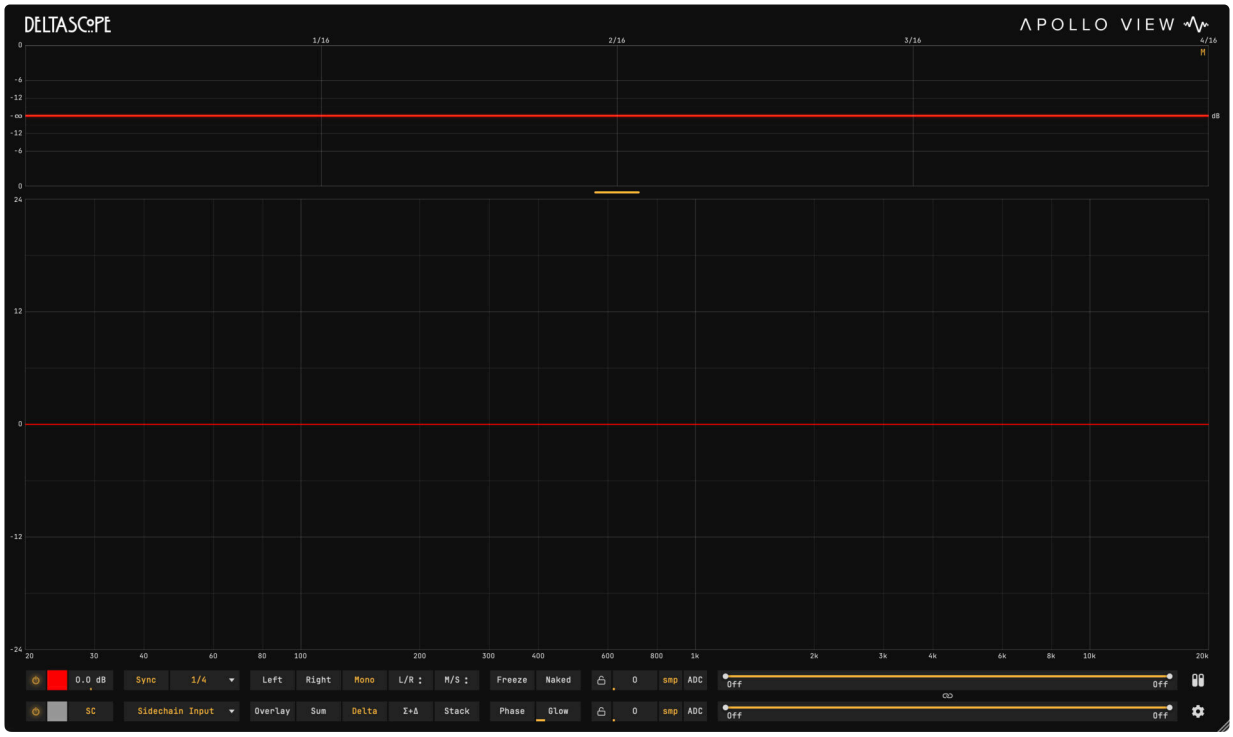
Difference view: the spectrum of everything the processing is changing.

TRACING A PROCESSOR'S REAL CURVE

Any processor between the two points where you take your signals can be traced, as long as it behaves like a filter: it changes the level and timing of each frequency, and adds nothing new.

1. Put the processor on a track and insert DeltaScope after it.
2. Give DeltaScope the signal from before the processor. The simplest way in any DAW is a second DeltaScope inserted ahead of the processor on the same track, picked from the source picker (a pre-FX send also works in DAWs that offer one, such as REAPER).
3. Click `Delta`.
4. Open Settings and set the spectrum's `View` to Magnitude.

DeltaScope now draws the frequency response of whatever processing is between the two signals: an EQ's actual curve, a console emulation's tilt, an entire chain, a signal sent out to hardware and back, drawn from your own material as it plays. You never need to open the processor's UI.



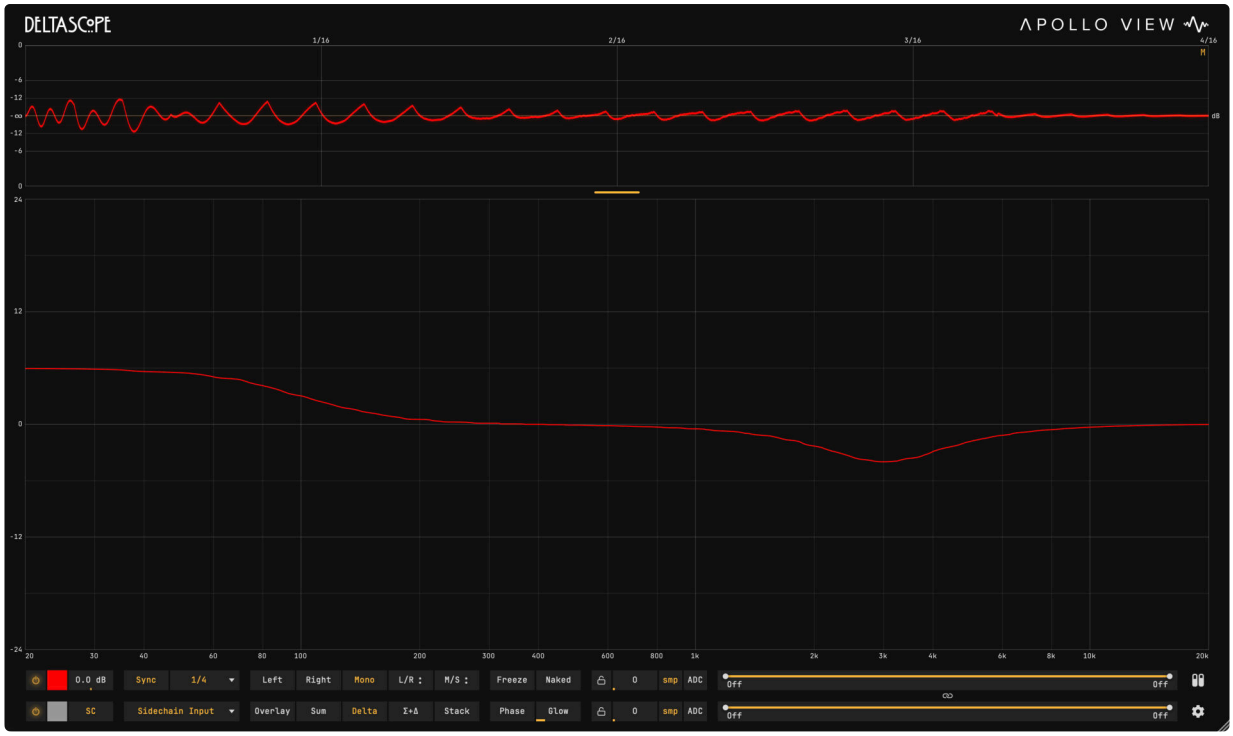
The baseline: identical signals draw a flat line at 0 dB. Anything that moves this line is something the processor is doing.

Read it against the fixed ±24 dB axis: 0 dB means untouched, above is boost, below is cut.

- If the Magnitude line stays flat while the Difference view shows a lot, the processing is changing phase or timing, not tone. That's a real result, not a fault in the measurement.
- A compressor is different again. It moves the whole line away from 0 dB (below it, unless make-up gain raises it), and the line rises and falls with the music. That's because Magnitude compares the levels of the two signals over the last stretch of audio.
- One limit: the curve can only be measured where your material has energy. Where the source has no energy at a frequency (for instance, the highest frequencies of a mix with little high-frequency content), there's nothing to compare. The line settles at 0 dB there,

which reads as “untouched” whether or not it is. Play something broadband through the chain to trace the whole range.

TIP: Magnitude works on your own material, so a test signal is optional. To see a processor’s complete curve rather than only where your music has energy, play broadband noise through it for a few seconds: pink or white noise from your DAW’s signal generator (Logic’s Test Oscillator, Pro Tools’ Signal Generator, or any noise plugin), inserted ahead of the processor. White noise measures every frequency equally well; pink favours the low end. Magnitude is a ratio, so the level doesn’t matter, only that every frequency has something to measure. Bypass the noise before you bounce.



A hidden EQ, traced live on programme material: a +6 dB low shelf at 100 Hz and a 4 dB dip at 3 kHz, read straight off the display.

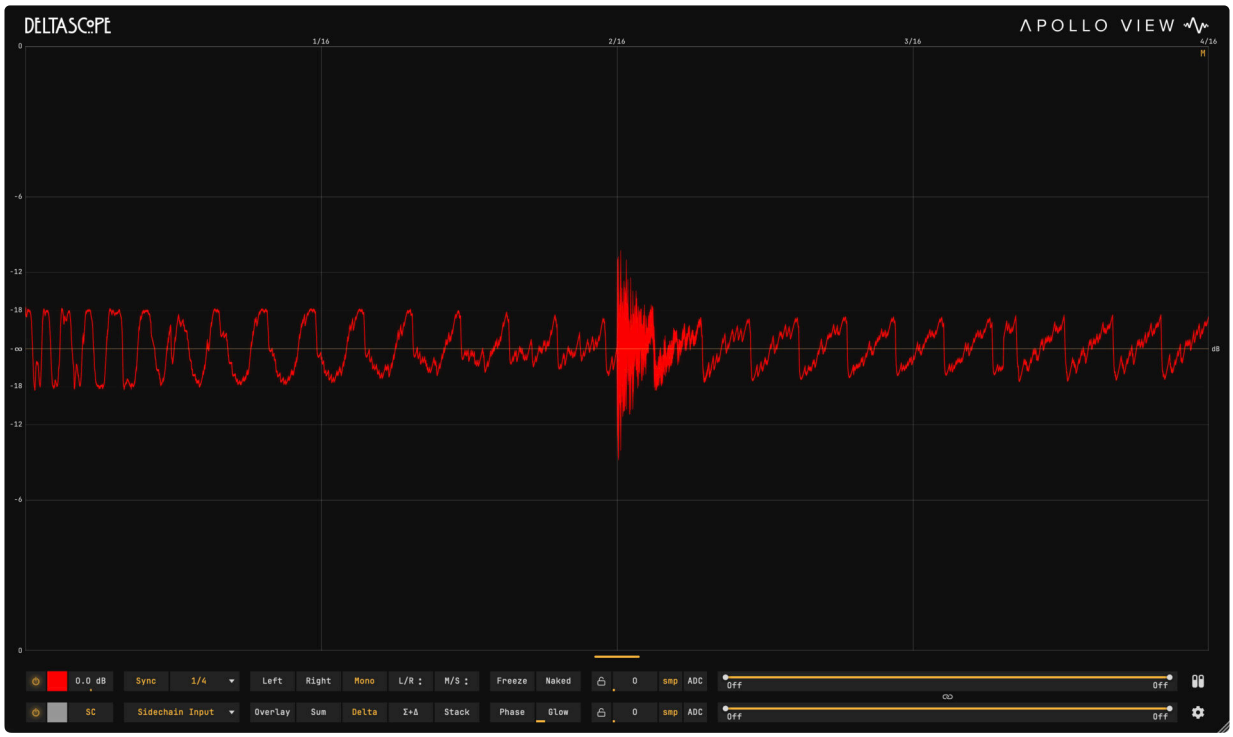
HEAR THE DELTA

A level-matched A/B asks whether a change sounds better; soloing the difference asks what the change is. A loud delta isn’t proof of an audible difference, and a quiet one isn’t proof of transparency.

With `Hear the Delta` on (Settings > General, on by default), choosing the Delta comparison mode switches the plugin’s output to the difference signal. What you hear is the delta lane’s signal, before the display gain and any band filter:

- In Left, Right or Mono you hear that one delta on both outputs.
- In L/R, you hear the L and R deltas.
- In Mid-Side the left output carries the mid-delta and the right the side-delta, so you’re hearing two analysis channels side by side, not a stereo image.

Click any other mode and the output is the unchanged track again.



What the saturator is changing. With Hear the Delta on, the delta on screen and the audio at the output are the same signal: harmonics, transient bite, hiss.

For a moment after you link a source, and again at each play start while the linked read has not arrived yet (for up to half a second), the delta lanes (and the output, with `Hear the Delta` on) are silent while DeltaScope aligns the two signals; otherwise you would see and hear a full-volume copy of the track. The scope shows the same silence; the lanes return aligned.

With any sidechain, `Hear the Sum` and `Hear the Delta` play 5 ms behind the live track, so they can fade through silence when you switch them. With another DeltaScope instance as the sidechain they can play further behind:

Source	Extra delay
`Sidechain Input`	None.
Another instance that sits earlier on the same track, when the host never runs it late	None.
Any other linked instance	At least 20 ms, rounded up to whole audio buffers: about 21 ms at 48 kHz with buffers up to 1024 samples, about 43 ms at 2048.

The track itself passes through with no latency.

> **CAUTION:** With `Hear the Delta` on (the default), choosing `Delta` replaces the plugin's output with the difference signal, and with `Hear the Sum` on, choosing `Sum` replaces it with the two signals added. Every other comparison mode, `Σ+Δ` included, leaves the audio untouched. If DeltaScope is on your master bus, what you hear or record in those modes is the monitored signal, not the mix.
>
> - An offline bounce is the exception: it always passes the track through untouched, whatever the monitor switches say, and the display rests until playback returns.
> - A real-time bounce is ordinary playback and prints the monitored signal.

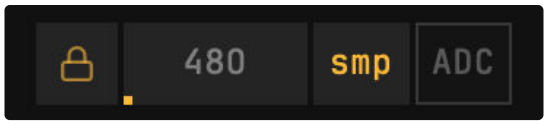
HEAR THE SUM

`Hear the Sum` (Settings > General, off by default) does for Sum what `Hear the Delta` does for Delta: with it on, choosing `Sum` sends the two signals added together to the output, aligned like the delta. On its own it auditions a sidechained part through the track it's compared with.

With `Hear the Delta` on too, the `Sum` and `Delta` buttons become a polarity switch: Sum is one signal plus the other, Delta one minus the other, so clicking between them flips the second signal's polarity and nothing else. Fix kick and bass phase, under Use cases, is a quick polarity check by ear.

ALIGNMENT: OFFSET AND ADC

A comparison depends on good alignment, and a null test depends on it most. If main and reference are even one sample out of time, the high frequencies stop cancelling and a quiet high-frequency residue appears where there should be silence. A few hundred samples of plugin latency turn the null into a comb filter: the two copies cancel at some frequencies and reinforce at others, leaving an unsteady residue instead of silence. So each signal row has its own alignment cluster:



The alignment cluster: padlock, offset, units and ADC, here locked at 480 samples after an ADC pass.

- **Sample offset slider** (the offset box): tells DeltaScope how late that row's signal arrives, in samples or milliseconds (units button). That row is shifted earlier to compensate, in the picture and in the comparison. Click to type an exact value; Alt-click or double-click resets to 0.
  - **Range**: up to one second. An ADC reading on an unlinked row can go further: it waits up to five seconds for a hit, so it can set up to five seconds, as long as that stays within 262,143 samples, the most the Sum and Delta lanes apply (2.7 seconds at 96 kHz). On a linked pair ADC stops at one second. At sample rates above 192 kHz the limit for every offset is that same 262,143 samples (0.68 seconds at 384 kHz), and ADC listens to a shorter stretch of audio.
  - **While you listen**: with `Hear the Sum` or `Hear the Delta` on, a drag bends the sound to the new offset rather than clicking at each step, and a settled null is exact. A move of more than 256 samples (a large typed value, an ADC result, a drag too fast to follow) goes through a short silence until the drag slows right down or rests. A smaller move slides, as a drag does.
- **Calculator**: the offset box does the adding for you. Where Latency between two linked instances (below) says to type a latency (in Ableton Live, or for a plugin that doesn't report its delay), read the latencies your DAW reports for the plugins in the chain and type them as a sum, `256+128+64`; Enter sets the total. It subtracts, multiplies and divides too, with brackets where you need them: `1024-256`, `(64+32)*2`, `2048/4`. In milliseconds it works the same way, `5.8+2.9`.
- **Lock** (padlock): protects the offset from accidental drags, and greys out that row's `ADC` button along with the slider, so a settled offset can't be re-measured by mistake either. It engages on its own ten seconds after you stop adjusting, and the moment you type a value into the slider. If `ADC` is greyed out after a small adjustment, the padlock has engaged: click it to unlock, and the slider and the button are active again.

LATENCY BETWEEN TWO LINKED INSTANCES

Plugins between the two linked instances add latency. What you type depends on your DAW:

Host	What to type
REAPER, Logic, WaveLab, Renoise, Cubase, Pro Tools, Bitwig and LUNA	A plugin between the two instances that reports its latency is already lined up for you: leave the offset at 0 and the Delta nulls. Type its latency as well and you undo the null. A plugin that doesn't report its delay still needs that delay typed into the offset.
Ableton Live	Type the latency of the plugins in between.
Other hosts	If the Delta doesn't null at 0, type that latency.

ADC ON ONE INSTANCE

`ADC` (auto delay compensation): each signal row has its own, on the main row and on the sidechain row. On a linked pair it works differently (see ADC on a linked pair).

1. Click `ADC` to arm it.
2. Place the play cursor right on a clear hit that starts from silence. A kick on the downbeat of bar one, played from bar one, is ideal.
3. Start playback.

DeltaScope counts the samples from the moment the transport starts to where the first sound begins in that row's signal, sets the row's offset to that count and locks it. If playback starts at bar one with the hit on the downbeat, that count is the chain's latency. Start mid-song and it only counts to the next loud hit, so cue the transport to the hit itself first.

It finds the first sound loud enough to count (louder than −40 dBFS), then steps back to where that sound rose out of silence, as far as −80 dBFS, if it did so within the 10 ms before. A sine kick therefore reads its first sample, not a few dozen samples late. Some sounds read later:

- A pad or attack that takes longer than 10 ms to rise from −80 to −40 dBFS reads at the first of −80, −70, −60 or −50 dBFS that it crossed within the 10 ms before the hit, or at the hit itself if it crossed none.
- A noise bed, bleed or a long fade that puts the quiet start more than 10 ms earlier can leave the reading on the hit, a little after the true start.

For a reading that is exact by construction, play a click. A click reads as it did before 1.4.0. When the plugins before DeltaScope are what you want measured, put a second instance in front of them and link the two, which is exact through the chain.

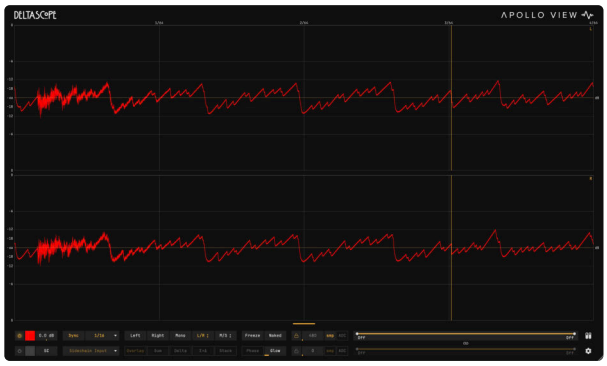
Two messages in the title bar say why a reading was not made:

Message	When it shows
`ADC: no hit in five seconds. Cue a hit from silence, then arm.`	Nothing loud arrived within five seconds, so the button disarms on its own. Stop, cue the transport to a louder hit that starts from silence, and arm again.
`ADC: the offset would pass the limit at this sample rate.`	The Sum and Delta lanes cannot apply the reading (more than 262,143 samples, or one that would leave the two rows further apart than that). It is not written: both rows stay as they were.

ADC is also useful with no reference at all. Plugins inserted **before** DeltaScope add latency, so in Sync mode (see Timebase) the whole waveform is drawn to the right of the beat grid. Arm `ADC`, play, and the display realigns: on drums, the downbeat kick moves back to the left edge of the scope, on the grid line.



A chain with latency ahead of DeltaScope draws the downbeat to the right of the grid line: 480 samples of plugin delay, visible.



ADC armed, played, measured: the kick back on the left edge, offset set and locked.

OFFSETS ON A LINKED PAIR

A sidechain row linked to another DeltaScope takes that instance's main offset: its offset box is greyed and shows the number, and you change it in the linked instance.

The main offset box of the instance that receives shows that row's whole offset: the linked instance's number plus the difference between the two. What you type there, or what `ADC` finds, sets the difference. When the linked instance's offset changes, both rows move together, the main box follows by the same amount and the Delta stays nulled. The main row may be the earlier of the two, so its box can read less than the greyed one.

An example in Ableton Live
An instrument reports 484 samples, followed by DeltaScope, your plugins, then a second DeltaScope linked to the first. Live hands both instances the sound 484 samples late, and DeltaScope cannot know that.

1. Type 484 once, in the first instance's main offset. The second shows 484 in both boxes and both waveforms sit on the beat lines.
2. If a plugin of 128 samples sits between them, open the main padlock, then press `ADC` in the second, or type 484+128 there. It reads 612.

A chain of three instances
A row takes its number from the instance it is linked to, and no further. Take three instances: the second is linked to the first, and the third is linked to the second.

Instance	Linked to	Its sidechain offset box
Second	The first	Takes the first's main offset.
Third	The second	Takes nothing, because the instance it is linked to is itself linked. Its offset box stays live: type its offset there.

Padlocks
When a linked row starts taking a number other than 0 from the linked instance, both offset padlocks of the instance that takes the number, the receiving one, close: main and sidechain.

When	What happens
You pick the instance, the row is switched on, the linked instance starts passing a number, or its number leaves 0	Both padlocks of the receiving instance close.
The row takes 0	Nothing closes.
You open a padlock afterwards	It stays open while the linked instance's number changes.
You are typing in or dragging the main offset box when the number arrives	The padlocks close when you finish.
A project, an undo or imported settings open	The padlocks are as saved.
The main padlock is closed	It greys the main box and its `ADC` button, as always.
The row stops taking a number: the linked instance is removed, or you choose `Sidechain Input`	Its box keeps the last number and the padlocks stay as they were, so open the sidechain padlock to type in it.

Reopening a project
A project with linked instances reopens with every offset and padlock as saved, whatever order your DAW loads the plugins in.

If the instance a row was linked to is not in the project when it opens (for example, the track came from another project), the row waits and the source list reads `Sidechain Input`. Choose a source from the list, `Sidechain Input` included, and the row works as any other.

`Import Settings...`, `Restore Defaults` and `Factory Reset` end the wait too. Instances linked to a waiting one keep their own numbers.

A link whose instance had already gone when the project was saved (its track was deleted, and the project saved after that) does not wait: it acts as an unlinked row at once. The source list reads `Sidechain Input` here too.

Projects from 1.3.0
A project from 1.3.0 keeps the line-up between its two rows and its Delta null. Where a linked row had its own typed offset that differs from the linked instance's main offset, both rows open that far from the beat lines. To put them back, type that row's offset from 1.3.0 into the linked instance's main offset. 1.4.0 no longer shows that number, because the row's box now shows the linked instance's offset, so if you don't have it, change the linked instance's main offset until both rows sit on the beat lines.

ADC ON A LINKED PAIR

On a linked pair, `ADC` reads the delay between the two instances.

1. Put one DeltaScope before your chain and one after.
2. Link the second one's sidechain row to the first ("Another DeltaScope instance").
3. If the main padlock is closed, open it.
4. Press `ADC` on the second one's main row.
5. Play from the top for an exact reading.

It reads in one of two ways:

Reading	When, and what it does
From the first sound	When the start is clean. DeltaScope takes the first sound to reach each instance and sets the main row's offset to the difference (plus the sidechain row's offset, which on a linked row is the linked instance's main offset), to the sample, within a fraction of a second of that sound and straight through compressors and the like, because the first sound arrives before they act. Both buttons go out.
From the music	When you press `ADC` while playing, or when the start is not clean: sound already at the top of the main input, a noise bed, a fade, or a chain that moves the first sound. It compares what arrives at the second instance with what the first one sees, and no test tone or clear hit is needed. After a few seconds of playback it sets the main row's offset to the delay of whatever sits between the two instances, locks it, and both buttons go out. The sidechain row is left alone.

When the reading is not made the title bar says why, for ten seconds or until you press `ADC` again:

Message	When it shows
`ADC: no reading. Arm, then play from the top.`	It gave up.
`ADC: the offset would pass one second.`	The offset would pass the range. Above 192 kHz, where the Sum and Delta lanes' limit is the smaller, it reads "ADC: the offset would pass the limit at this sample rate." instead.
`ADC: this track is ahead of the linked one. Set the linked instance's offset first.`	The linked instance's own offset is larger than the reading allows.
`ADC: the linked track runs too far behind at this rate. Use a smaller buffer, then arm.`	The linked track runs further behind than the search can reach, which happens when the host sends bigger blocks than it announced.

It gives up, buttons out and nothing changed, on unrelated audio, silence, a pure tone, pitch shifting, an offset that would pass one second, or when it can't be sure within about eight seconds. Silence between hits is fine. Gain, polarity, EQ, compression, saturation, chorus and reverb mixed with dry signal don't stop it, and neither do loops and drum-machine material.

Some limits:

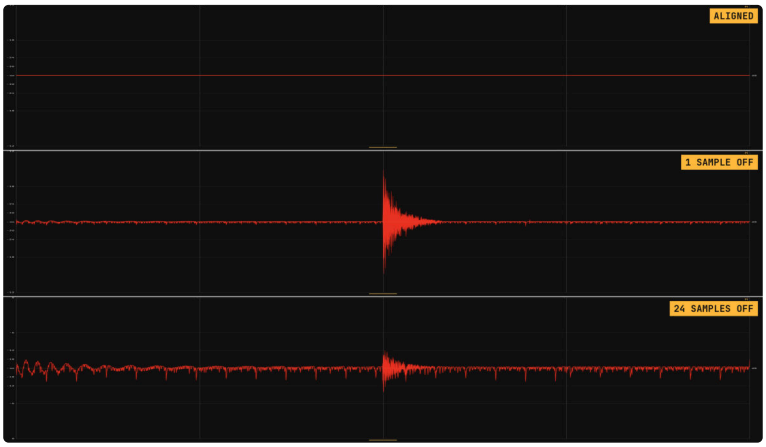
- Through a compressor the reading can sit a fraction of a millisecond early, so check it by eye.
- A sustained bass that repeats exactly may not lock.
- When a loop repeats exactly, a delay longer than the loop's own repeat reads short by a whole number of repeats (half a second for a four-on-the-floor kick at 120 BPM). Anything that varies in the loop, a bass line, a fill or a vocal, removes that limit.
- If the transport stops, the button stays lit and the search starts again on play.
- Content that lies wholly below a steep low-pass can lock at the filter's own group delay, which is a real delay of that content.

On the sidechain row of a linked pair:

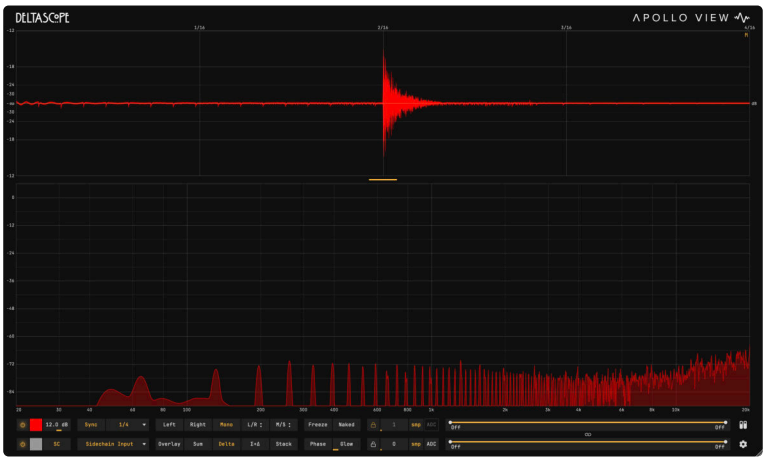
- `ADC` is greyed while the row is linked, because the pair has one reading and the main row's padlock guards it.
- The row's offset box is greyed too: it shows the linked instance's main offset, and to change that number you change the main offset in the linked instance (the box's tooltip says so).
- The exception is a row linked to an instance that is itself linked: no number is passed there, so its box stays live and you type the row's offset in it.
- The greyed box and the button come back when the row returns to the host's `Sidechain Input`.

A sidechain row that takes a host bus is unlinked, so it keeps the clear-hit method in ADC on one instance.

FINISHING BY EYE



Aligned, one sample off, 24 samples off: the same null with the offset moved. ADC measures the chain's latency for you, but a noise bed or a slow fade can leave the reading on the hit rather than on the first sound. Unlock, adjust the offset a sample at a time, watch the delta flatten to a line, and let the padlock lock again on its own.



The spectrum of a one-sample error is a clean ramp rising towards the top of the range. The two signals still cancel in the bass, where one sample is a tiny fraction of a cycle, and cancel less and less as the frequency rises. Remember the shape: it's the sign of a single sample of misalignment.

TIP: Treat ADC as the first step. Let it give you an approximate value, then:

1. Unlock the padlock.
2. Adjust the offset a sample at a time while watching the Δ lane.
3. Stop when the line goes flat.

Where the offset needs a typed latency (see Latency between two linked instances), your DAW's per-plugin latency figures are a way to get it: type them into the offset as a sum, such as 256+128+64, and Enter sets the total. That's only as good as the figures, since not every plugin reports its latency correctly. On a linked pair in REAPER and the other hosts named there, don't add them: the DAW has already lined the pair up, and typing them undoes the null.

PHASE CORRELATION

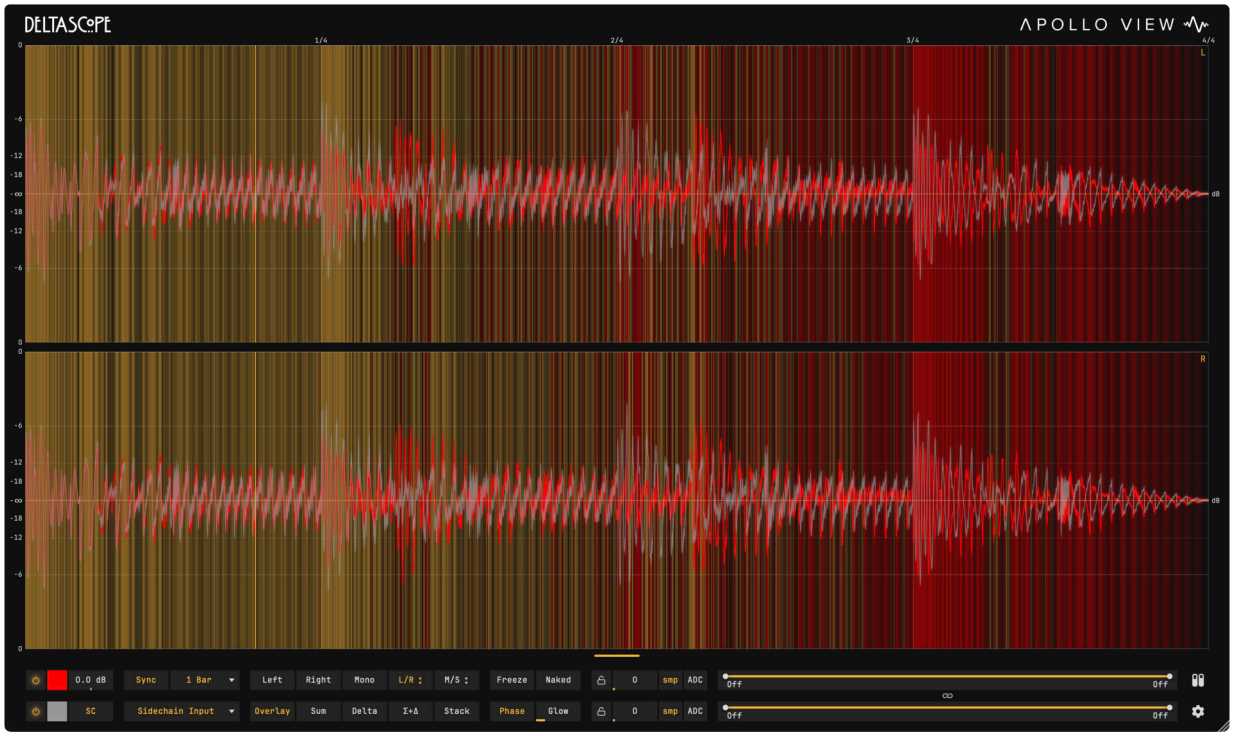
This is different from the correlation meter in your DAW. That meter compares left with right, averaged over time. DeltaScope's phase display compares the **main signal with the reference** (kick against bass, processed against dry) and shows the result along the waveform, so you see where in time the two cancel, not only that they sometimes do.

`Phase` overlays the correlation on the scope, as bars or as a heat fill behind the waveforms. It's in the sidechain row because it needs a reference: until a reference (host sidechain or instance link) is connected and `SC` is on, there's nothing to correlate against.

+1 means the two move together (gold: they reinforce when summed); **0** means no consistent relationship; **−1** means they move in opposition, as if one had its polarity flipped (red: they cancel when summed).

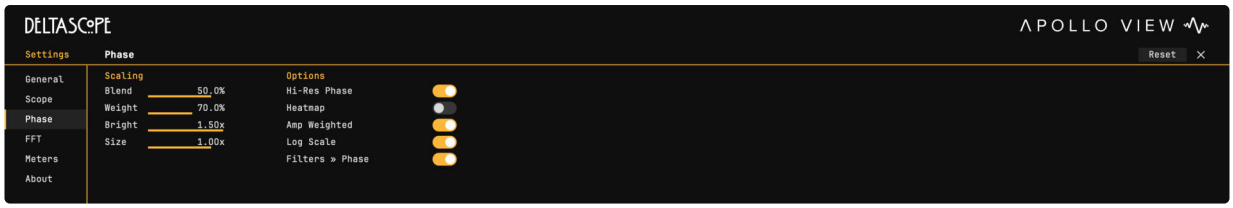


Correlation bars, main against reference: gold bars near +1 while the two are in phase, red bars near −1 as they move out of phase later in the musical bar.



The same drift in the Heatmap style: gold where the signals are in phase, red where they fall out of phase.

PHASE SETTINGS (SETTINGS > PHASE)



Settings, Phase page.

Section	Control	What it does
Scaling	`Blend`	What the display measures: 0% is pure correlation, which ignores level (“are these in phase?”); 100% is interference (“will these cancel when summed?”). Default 50%.
	`Weight`	How much more the louder moments count than the quiet ones while `Amp Weighted` is on, 0 to 200%: 0% treats every moment equally; 100% is normal weighting; 200% is stronger. Default 70%.
	`Bright`	How strongly the bars or heat fill glow. Turn it down when the overlay fights the waveform for attention, up when a quiet passage’s colours are too faint to read. Default 1.5 (0.25 to 6).
	`Size`	How tall the bars stand in the lane: short, they sit under the waveform as a quiet strip; tall, the correlation becomes the main event. Greyed in Heatmap style, which has no bars. Default 1.0 (0.25 to 4).

Section	Control	What it does
Options	`Hi-Res Phase`	Twice as many correlation bars across the display, 64 per grid division instead of 32, so shorter moments get their own bars. The trade is a busier overlay. Default on.
	`Heatmap`	A continuous colour field instead of bars: trouble shows at a glance across a whole passage, while bars resolve individual moments. `Size` doesn't apply. Default off.
	`Amp Weighted`	Weight the display by amplitude so quiet passages don't show strong readings that mean nothing. Default on.
	`Log Scale`	Draws bar height and heat strength on a dB scale instead of a linear one, so weak correlation stays visible rather than vanishing near zero. Default on.
	`Filters » Phase`	Band filters also apply to the correlation analysis (see Band filters, the next chapter). Default on.
	`Reset`	This page back to its defaults.

BAND FILTERS

A picture is easier to read when it shows only the range you care about. When you check a null in the bass you don't need to see the cymbals, and what a saturator does to the mids is easier to see when the kick's fundamental is filtered out. So each signal row ends in a two-handle slider: a high-pass and a low-pass, 20 Hz to 20 kHz (the high-pass handle stops at 15 kHz, the low-pass at 30 Hz). The filters are zero-phase, so the filtering never shifts what you see in time. Move both handles to the ends of the range and the band is `Off`.



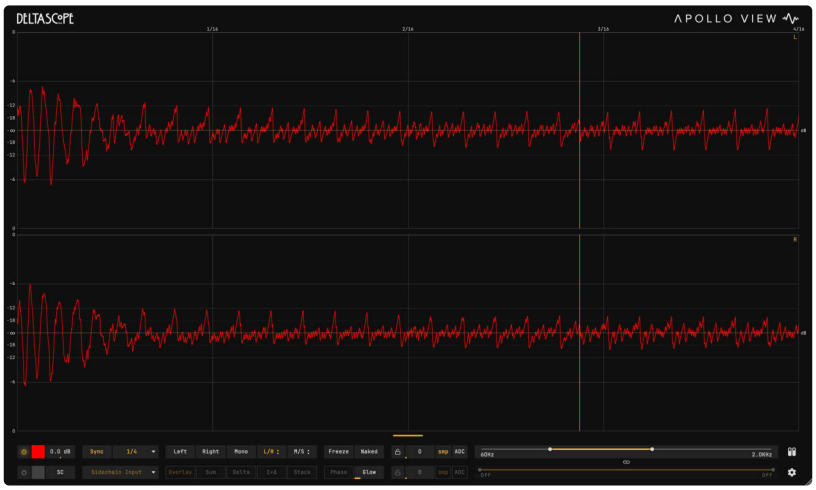
The band filter: high-pass and low-pass handles, with the link button beneath.

An analyser working on captured audio can filter with zero phase shift; a real-time EQ can't do this without adding latency. An ordinary filter shifts the frequencies near its cutoff slightly in time (that shift is "phase shift"), which would move the waveform on screen. DeltaScope's filters run over the captured stretch of audio forwards and then backwards. The two time shifts cancel, so nothing moves. That's why offset readings, nulls and phase checks made through the band stay valid.

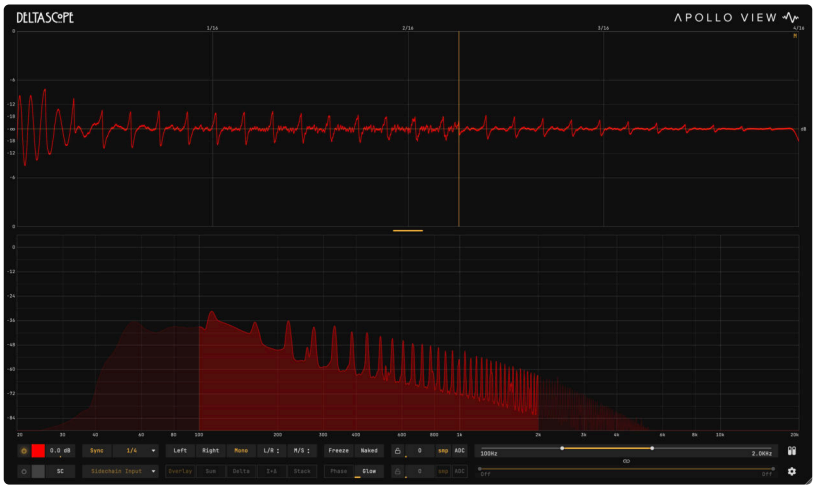
The coloured bar between the two handles is a handle too. Drag it and the whole band slides along, keeping its width: a quick way to sweep a fixed-width band across the spectrum. It's useful with Delta: sweep the band while watching the delta to find the frequency range a processor changes.

> **CAUTION:** The band filters are purely visual. They filter what DeltaScope analyses and draws, never the audio you hear. The signal passing through the plugin stays untouched wherever the handles are.

The link button between the rows ties both filters together, so main and reference always compare the same band. Two switches in Settings apply the band to the spectrum (`Filters » FFT`) and the phase display (`Filters » Phase`); in the spectrum the out-of-band region dims, so you can see which range is being analysed. The Magnitude view is the one exception. It always compares the two full-band signals, because a filtered ratio would no longer be the processor's curve.



Main band 60 Hz to 2 kHz: the scope draws only the range in question; the audio through the plugin is untouched.



Band at 100 Hz to 2 kHz with Filters » FFT active: the spectrum fill dims outside the band.

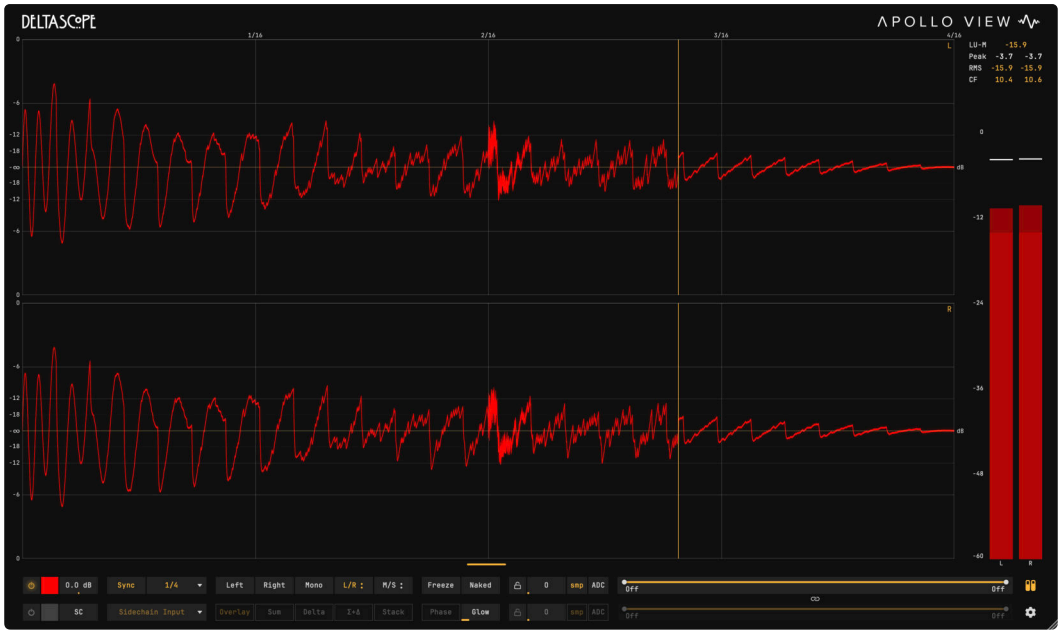


Band at 100 Hz to 2 kHz applied to the phase display: correlation is analysed inside the band only.

LEVEL METERS

The meters button, at the right end of the main row, docks a meter strip beside the display, directly above the button:

- **Peak bar**: one per channel (L/R, or Mid/Side in M/S mode). Sample peak reads the loudest sample. `True Peak` (the oversampling filter of ITU-R BS.1770-4: 4x at 44.1 and 48 kHz, 2x at higher rates) reads the peaks that fall between samples. A converter or a lossy encoder can produce those peaks, which is why delivery specs ask for a true-peak figure. Ceiling, floor, release and peak hold are configurable. The bar is red above 0 dBFS, and the held line turns red while it is at or above 0 dBFS.
- **Secondary bar**: per-channel RMS, with its window set in Settings > Meters.
- **Readout**: loudness as streaming services and broadcasters measure it (BS.1770), as a momentary (LU-M), short-term (LU-S) or whole-programme integrated (LU-I) figure, alongside per-channel peak, RMS and crest factor (CF).
  - The peak figure holds its highest value for two seconds before it decays, and turns red while that channel's held peak is at or above 0 dBFS.
  - Double-click the loudness readout to restart an integrated (LU-I) measurement. Hiding the meters with their button starts it over too, and starts the momentary (LU-M) and short-term (LU-S) readings from nothing when you show them again, since nothing is measured while they're hidden; opening Settings or Naked mode doesn't.
  - Above 192 kHz the Short-term figure averages less than 3 seconds.
- **Crest factor**: the gap between peak and average level, how much dynamic range is left; a smaller number means a more compressed signal. The readout holds its lowest recent value, the most compressed moment, for two seconds before releasing, so a limiter that is compressing too much appears as a low number. The RMS window is also used as the crest-factor window.



Meter strip docked: peak and RMS per channel, programme loudness on top.

In Mid-Side mode the bars and the peak, RMS and crest-factor readings all measure mid and side. The one exception is the loudness figure at the top: it stays the loudness of the whole

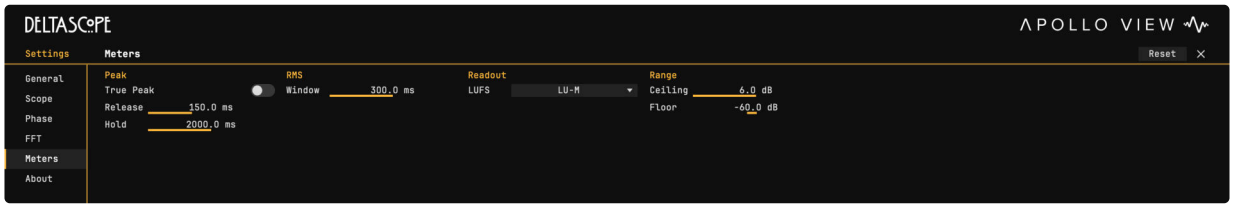
programme, not of mid or side on their own, because that's what the standard defines and what a streaming service will measure. Switching channel modes never changes your loudness number.

Two more readings can look wrong but are correct:

- Programme loudness sums both channels, so a centred signal reads about 3 LU above what either channel alone would.
- `True Peak` can legitimately read above 0 dBFS; finding inter-sample peaks is the purpose of the measurement.

For reference, a sine's crest factor is about 3 dB and a square's about 0 dB. `Freeze` stops the meters at the frozen moment along with the display (see Freeze).

METER SETTINGS (SETTINGS > METERS)



Settings, Meters page.

Section	Control	What it does
Peak	`True Peak`	Read the peaks that fall between samples (ITU-R BS.1770-4 true peak) instead of the loudest sample. Default off.
	`Release`	How quickly the peak bar falls back after a peak: short and the bar dances with every hit, long and it eases down for an easier read. (The held line above it is `Hold`'s job.) Default 150 ms.
	`Hold`	How long the peak bar's held line stays before dropping. Default 2000 ms.
RMS	`Window`	How much recent audio the RMS and crest-factor figures average over: short follows the music moment to moment, long gives a steadier number that's easier to compare. Default 300 ms.
Readout	`LUFS`	Loudness readout mode: LU-M (momentary), LU-S (short-term) or LU-I (integrated). Default LU-M. Double-click the readout to restart LU-I. Above 192 kHz LU-S averages less than 3 seconds.
Range	`Ceiling` / `Floor`	Meter scale in dB. Default +6 / −60.
	`Reset`	This page back to its defaults.

USE CASES

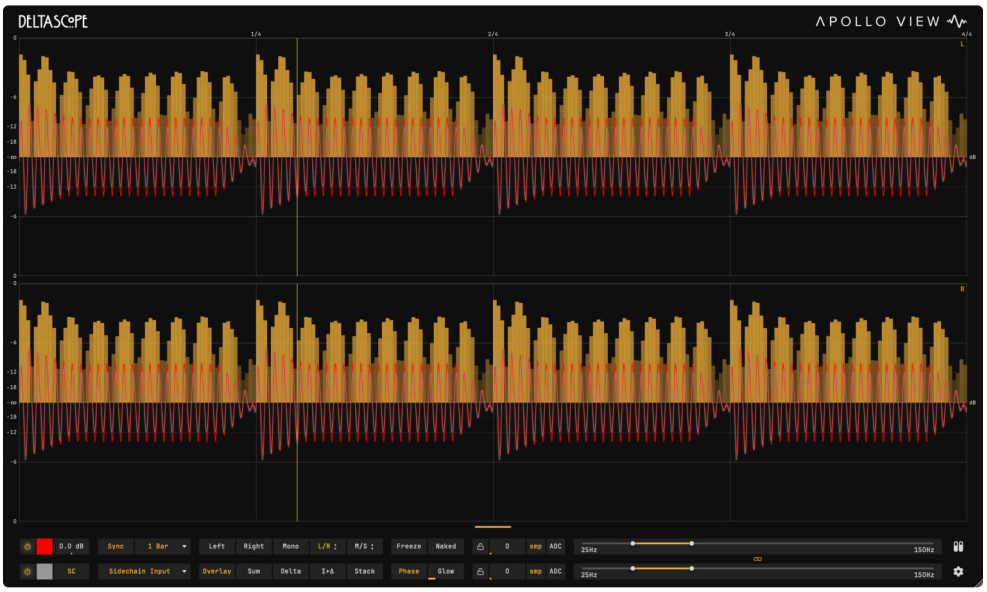
FIX KICK AND BASS PHASE

Kick and bass share the low end, and when they are out of phase the sum loses level in the overlap. EQ can boost the cancelled region, but it can't undo the cancellation.

1. Put DeltaScope on the bass bus and use the kick as the reference (sidechain or instance link).
2. Use the band filters to isolate the overlap region (roughly 25 to 150 Hz) so the analysis covers only that range.
3. Enable `Phase`. Correlation dropping towards −1 on the kick hits means the two are cancelling; `Overlay` mode shows the same thing in the waveforms.
4. Fix it on the bass with a polarity flip or a small time shift, until the correlation reads positive through the kick hits on the notes you care about. Then listen.



The cancellation, located. With the band at 25 to 150 Hz, the phase display compares bass against kick: red for the whole musical bar means the two are cancelling in the band where they overlap. A stereo correlation meter can't show this; it compares left with right, not your kick with your bass.



After the fix: same bar, same instruments, gold. Kick and bass now add together on every hit instead of cancelling each other.

PRO TIP: A quick polarity check by ear, with the roles swapped.

1. Put DeltaScope on the kick track, sidechain the bass into it, and turn `Hear the Sum` on (`Hear the Delta` is on by default).
2. Take the bass track out of the mix without cutting what it sends to the sidechain (how depends on the host: see below). The kick track is now the only place you hear the bass, and DeltaScope decides how.
3. Click `Sum` and you hear kick and bass as they are; click `Delta` and you hear them with the bass's polarity flipped. Go back and forth while the loop plays and keep whichever has the fuller low end; the scope draws the same two signals adding or cancelling as you listen.
4. If Delta wins, flip the polarity on the bass track, or nudge its timing until Sum wins. Then bring the bass back into the mix and set the mode back to `Overlay` .

Taking the bass out of the mix:

- **Logic Pro**: mute the bass track. The sidechain still carries it.
- **Ableton Live**: switch the bass track off with its Track Activator, and leave the second chooser under `Audio From` in DeltaScope's sidechain at `Post FX` , which is the default, or `Pre FX` . At `Post Mixer` the sidechain goes quiet with the track.
- **REAPER**: a muted track sends nothing, so don't mute it. Untick `Master send` in the bass track's routing window; its send to the kick track carries on.
- **Other hosts**: watch the sidechain trace as you mute the bass. If it goes flat, the mute has cut the send too: unmute, and turn off the bass track's output to the master in its place.

The sum you hear matches your mix only when the bass reaches the sidechain at the level it plays in the mix. Logic's side chain and Live's `Post FX` and `Pre FX` taps take the bass before its fader, so the match holds with that fader at 0 dB. In REAPER the send follows the fader, so leave the send's own level at 0 dB.

Fixing it at the source is what our Align plugin is for.

SEE WHAT YOUR PROCESSING ACTUALLY DOES

1. Insert one DeltaScope before your processing and one after.
2. Link the second to the first by track name (or sidechain the clean signal into it).
3. Choose `Delta` mode.
4. Use ADC and a small manual offset adjustment, or, where your DAW needs it (see Alignment, Latency between two linked instances), the latency figures it reports, to align the two signals to within a single sample.

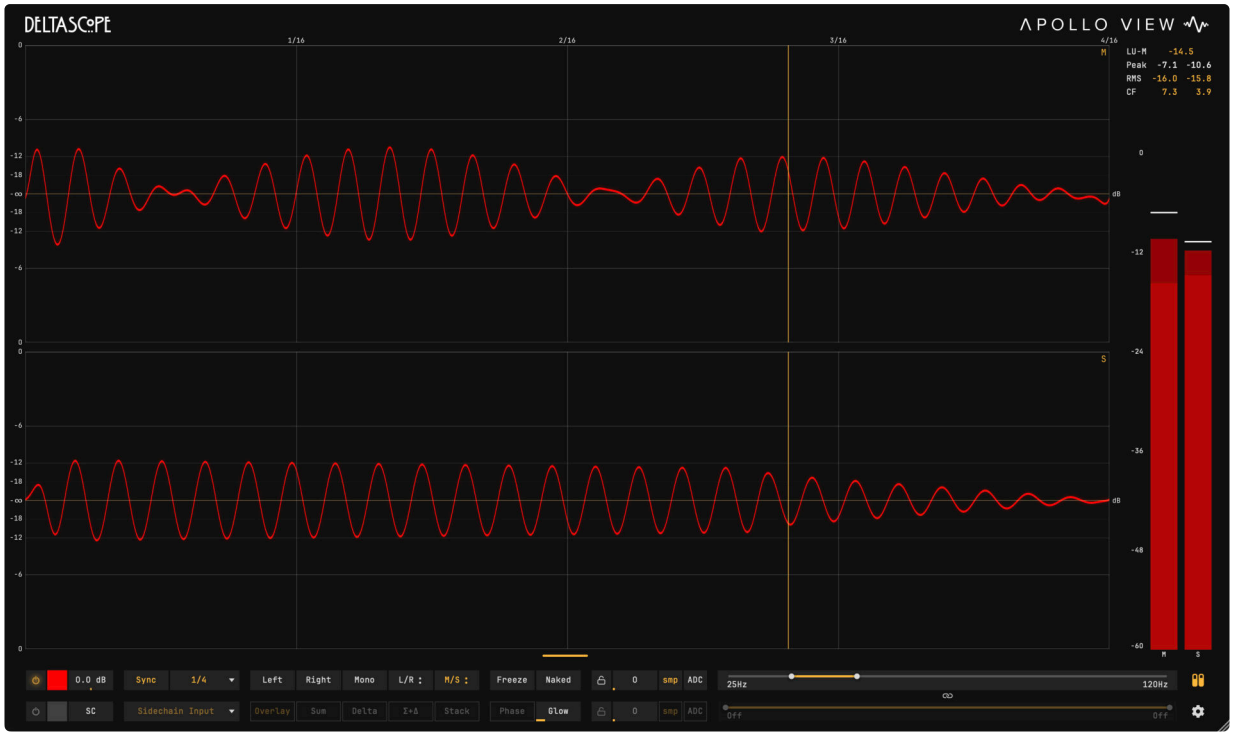
The scope draws what the processing is changing, and `View` set to Magnitude in Settings > FFT makes the spectrum show the processor's frequency response on your real material. Use it to check a Pultec-style EQ move, see what a compressor reshapes, or check a whole chain. The full walk-through is Tracing a processor's real curve, in Delta and Hear the Delta.

LISTEN TO THE DIFFERENCE

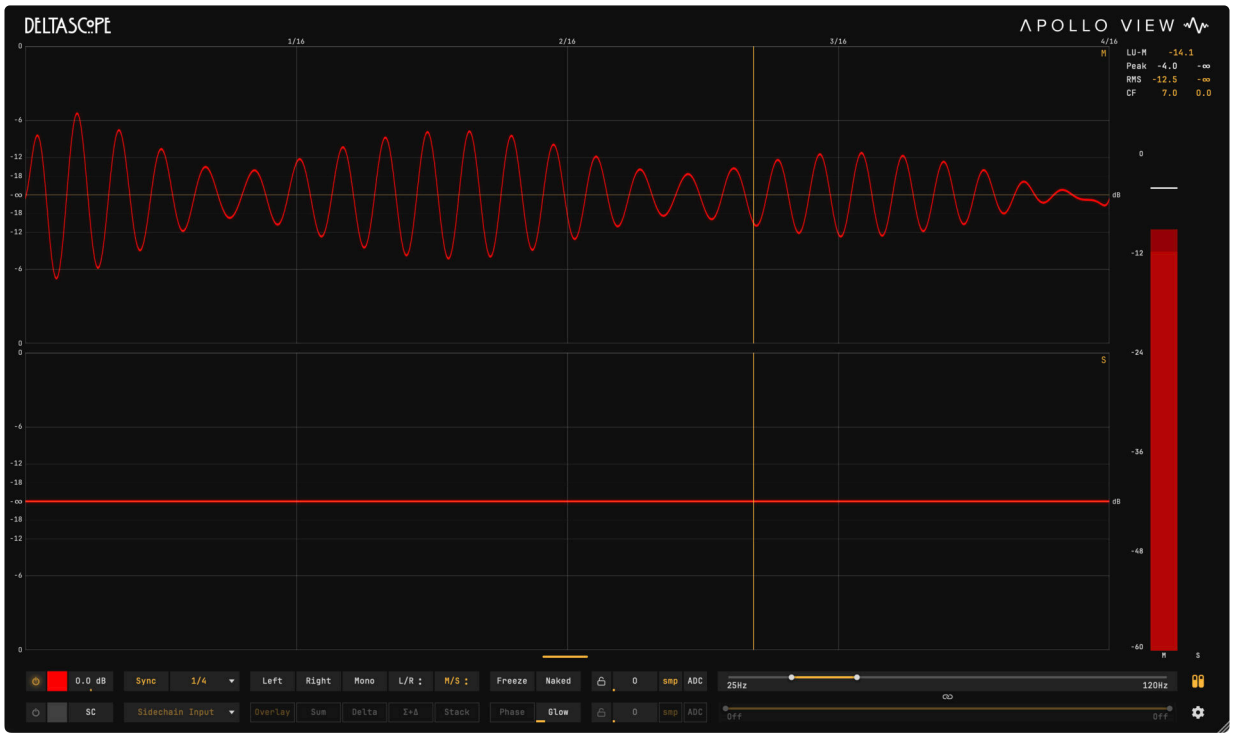
Use the same setup, but listen instead of looking. With `Hear the Delta` on (the default), DeltaScope's output **is** the difference signal for as long as `Delta` mode is selected. What you hear is everything the processing is adding or taking away, isolated: a limiter reducing transients, or what a plugin described as transparent actually changes. Remember that a delta you can hear on its own isn't the same as a change you'd hear in the mix.

CHECK WHAT SURVIVES MONO

`M/S` mode: anything that appears only in the Side lane disappears when the mix is summed to mono. Band-limited to the low end and look at the Side lane:



Band-limited to the sub and viewed in Mid-Side: the Side lane is carrying real low-frequency energy, and that part of the bass disappears the moment the mix is summed to mono. The meters read the full band, not the band you've isolated.

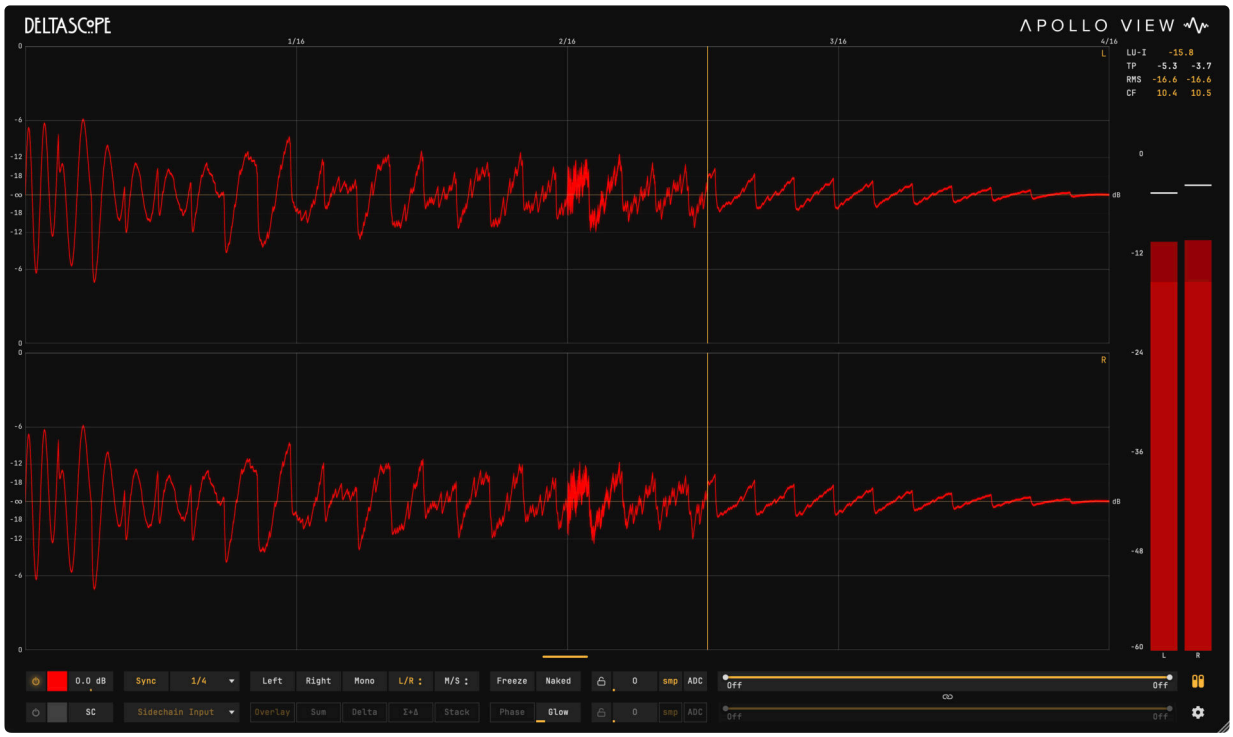


The same groove with a mono sub: the Side lane goes flat, and the low end survives mono.

For the correlation view of the same question, route the elements you're worried about into main and reference (kick against bass, double against lead) and watch where `Phase` turns red. Sweep the band filters to find which frequency range is cancelling.

DELIVER TO SPEC

1. Open the meter strip with the meters button and enable `True Peak` in Settings > Meters.
2. Set the readout to LU-I and double-click it to start a fresh integrated measurement.
3. In Settings > Meters, set `Hold` to its maximum (10 s) so the peak bar's held line stays long enough to read.
4. Play the track from start to end, watching the peak bar's held line through the loudest passages. At the end, the LU-I readout is your integrated loudness.
5. The highest value the held line reaches is your true-peak maximum; the gap between it and the spec's ceiling is your margin. The meters don't remember the highest peak of the whole programme, which is why you watch the held line as the track plays.
6. Compare both with the delivery spec you're mixing for. Most streaming services normalise to around −14 LUFS integrated and ask for a true-peak ceiling of −1 dBTP; broadcast specs are stricter. Those are the two numbers most specs ask for.



Deliver to spec: `True Peak` on (TP), LU-I integrated readout, RMS secondary bar.

MAKE VIDEO CONTENT

Naked mode hides every control and label, leaving waveform, spectrum or both, edge to edge: clean output for music videos, tutorials and livestreams. The hidden 16:9 lock (see Freeze and Naked mode) fits the window to a video frame.

SETTINGS

The cog opens Settings over the display. The title bar stays, so the tooltip strip keeps describing whatever you point at. The pages are listed down the left (`General`, `Scope`, `Phase`, `FFT`, `Meters` and `About`), and clicking a name opens that page. Each page lays its settings out in columns to fit the window at whatever size it is; only a window too small to show a page whole has to scroll, as the `FFT` and `About` pages do at the smallest size.

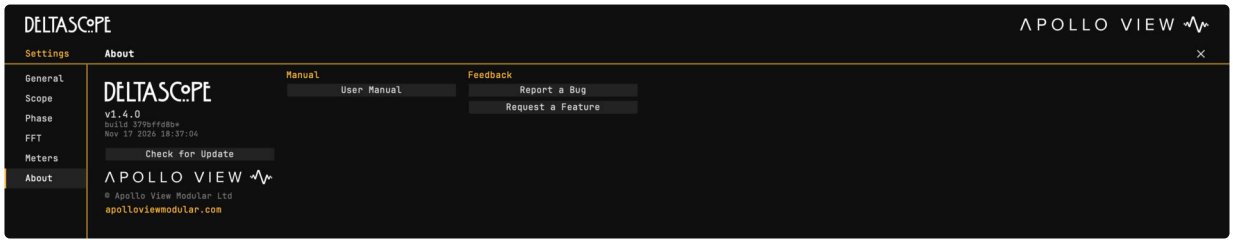
A page with a `Reset` shows it at the top right, and it returns that page, and only that page, to its defaults. Close Settings with the × at the top right or with Escape. Every change applies as you make it.

- **General**: everything that is about the plugin as a whole.
  - **Audio**: `Hear the Delta` is the delta monitor switch: with it on, choosing the Delta comparison mode sends the difference signal to the plugin's output (see Delta and Hear the Delta). `Hear the Sum` does the same for the Sum comparison mode, off by default; with both on, Sum and Delta are a polarity check (see Hear the Sum).
  - **Interface**: `Tooltips` turns the control descriptions in the title bar on and off. Below 1230 pixels wide the title bar shows none.
  - **Defaults**: `Save As Default`, `Restore Defaults` and `Factory Reset`.
    - `Save As Default` stores the current settings as the default for new instances. It's dimmed until a setting has changed since the last save or restore; if all you want is to save what you already have, change any setting and change it back. Some settings belong to one project or moment rather than to your general setup: Freeze, the sample offsets with their padlocks, and the instance link. These are not saved in the default, on purpose, so a new instance always opens with its offsets at zero and unlocked.
    - `Restore Defaults` loads the saved default into this instance, keeping its window size; it's dimmed until a saved default exists and something has changed since. Because a saved default holds no offset and no instance link, it also sets this instance's offsets, main and sidechain, to 0, opens both padlocks and ends the link: an alignment you have set on this instance is lost.
    - `Factory Reset` deletes the saved default and returns this instance to factory settings, window size aside. The offsets, padlocks and link go back to 0, open and none as well.
  - **Settings File**: `Export Settings...` writes this instance's current settings, as a project save holds them and not your saved default, to a file and `Import Settings...` loads one, so you can move your setup between machines or share it. An import takes everything but the window size, the offsets included, so check them after you import. Import checks the whole file and rejects a damaged one, or one from Scope, rather than half-applying it.
  - **Image**: `Export Image...` renders the current display (waveform lanes and spectrum) as a high-resolution PNG, laid out exactly as on screen and without the controls; a save dialog asks where. The image is drawn from the audio itself, not captured from the screen, so freeze first if you want a precise moment. If nothing has played yet there's nothing to export. Settings closes first: the display itself renders the image, and Settings sits on top of it.

    - **Licence**: shows the email the plugin is licensed to. `Deactivate Licence` removes the stored key from this machine at once and opens the activation screen (see Licensing), and `Licence Agreement` opens the licence on our website. Without a licence the section reads Demo mode and the button becomes `Activate Licence...`, which brings back the activation screen. If the stored key can no longer be used, it reads Licence key no longer valid, with the same button. While the licence is still being checked the section reads Checking licence… and the button works the same.
- **Scope**: `Clip Overlay` and `Clip Border`, `Cursor` and `Rolling` (the cursor has no meaning while the trace rolls, so it's dimmed then), and the `Colour by Frequency` section, whose `Main` and `Sidechain` switches colour the two traces (see Colour by frequency). If only one channel is by frequency, the knob shows in the middle with a note naming it; a click turns both on. This page has a `Reset`, which turns `Clip Overlay`, `Clip Border` and `Cursor` on, `Rolling` off, and both Colour by Frequency channels off.
- **About**: the version and, under it, a build stamp that distinguishes two builds of the same version; support will ask for both, and a bug report fills them in for you. `Check for Update` beneath them opens our website, where the newest version is. The gold address opens our website too. `User Manual` opens this manual in your browser, always the newest version. `Report a Bug` and `Request a Feature` open pre-filled support emails (the bug report fills in your plugin version, build stamp, OS, DAW, sample rate and buffer size for you).



Settings, General page.



Settings, About page: version and build stamp with `Check for Update` under them, the manual and the feedback emails.

WHAT IS REMEMBERED

Nearly everything you set is saved with the DAW project, per instance:

- Window size and the pane split, timebase, channel mode and whether L/R and M/S use one lane or two, colours, gain and Glow.
- The comparison mode and source, both band filters, both offsets with their units and padlocks.
- Every Settings page, the meter strip and `Phase`.

Reopen the project and it all comes back, links reconnected by track name. A few things are for the session only, on purpose, and start fresh every time:

- Freeze, the two power buttons, Naked mode, zoom and pan.
- An armed `ADC`, a measurement selection and the 16:9 lock.

A project never opens frozen, blank, zoomed or with its controls hidden.

FILES

DeltaScope keeps two files outside your DAW projects, both in `~/Library/Application Support/ApolloView/` on macOS and `%APPDATA%\ApolloView\` on Windows:

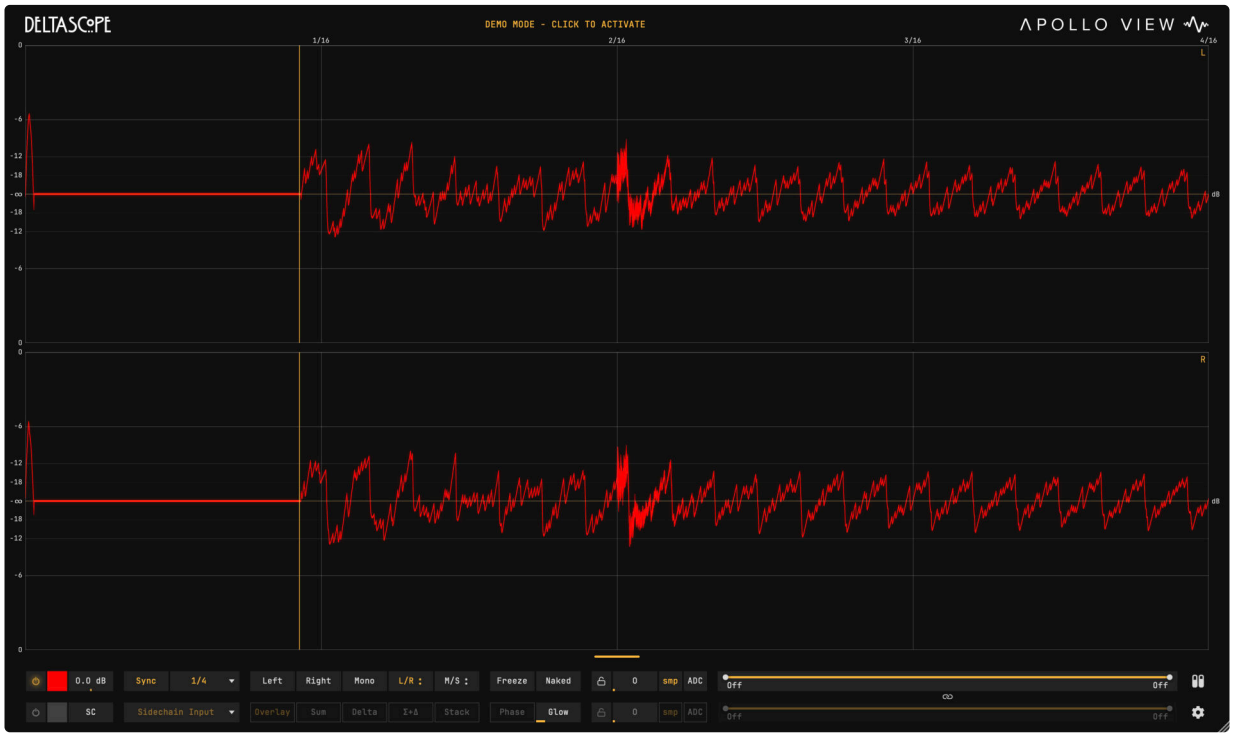
File	Back it up?	If you delete it
Licence: `DeltaScope.license`	No. Just activate again on the new machine.	The plugin returns to the activation screen the next time it opens.
Saved defaults: `DeltaScope/defaults.json`	Yes, when you move to a new machine.	New instances return to factory settings.

`Export Settings...` and `Import Settings...` in the panel's Settings File section move one instance's current settings as a single file, as a project save holds them. The export does not carry your saved defaults, so copy the defaults file to move those.

DEMO MODE

Without a licence, DeltaScope opens on the activation screen. Click `Continue in demo mode` and it runs as a full demo: every feature works. The activation screen comes back each time you open the plugin window. The display goes blank (the waveform flat, the spectrum, meters and phase display empty) and the audio mutes for about five seconds in every twenty-five, at a moment that changes from one mute to the next, with a gold `DEMO MODE - CLICK TO ACTIVATE` banner in the title bar. Clicking the banner reopens the activation screen.

The mute fades in and out without clicks, and every demo DeltaScope in the project mutes at the same moment, so a project with several of them goes quiet once each time rather than several times. An offline bounce is never muted: it passes the track through untouched, in the demo as in the licensed plugin. A real-time bounce is playback, so the mute applies there.



Demo mode: the gold banner in the title bar, and the flat gap a demo mute leaves in the waveform, with audio on both sides of it.

LICENSING

1. Buy a licence at shop.apolloviewmodular.com/plugins/deltascope (the activation screen's `Buy licence` link goes to the same place, and `Lost your key?` is the way to recover a lost key).
2. Open DeltaScope, enter your email and key, click `Activate`.

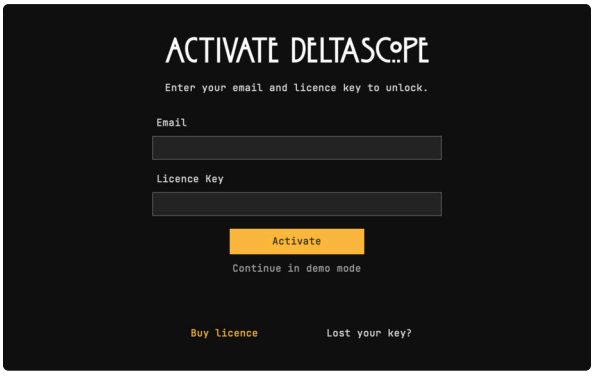
Activation is local. The key is checked on this machine, and once you have it no internet connection is needed; getting the key is the only online step. `Deactivate Licence` in Settings > General removes the stored key from this machine. To move to another machine, activate there with the same email and key.

If a key is not accepted, the activation screen says why and what to do:

- A key cut short or with extra characters: copy the whole key from your email or your account and paste it again.
- A Scope key.
- A key for another Apollo View plugin.
- A key for a newer version of DeltaScope, or for DeltaScope 2.
- A key that does not match the email you entered.
- A key from before 1.0: the screen says "Your licence needs a one-time re-activation for our upgraded security. Check your email for your new key."

A key that can no longer be used says so, and a `Contact support` link appears beside `Lost your key?`. Write to us and we will help.

If the activation screen reports that it activated but the licence could not be saved, the key was accepted and this session runs licensed, but the plugin couldn't write its licence file. Check the permissions on the ApolloView folder (both platforms' paths are listed under Files, in the Settings chapter of this manual), or you'll be asked to activate again at the next launch.



The activation screen, with `Continue in demo mode` under the button.

Licensing is designed so that a failed check never locks you out. If a licence check can't complete, the plugin keeps working. Once DeltaScope is running licensed, a licence file it can't read never turns it back to the demo: it retries rather than treating the file as missing. If the file can't be read when DeltaScope first opens, it plays in full for about three minutes and checks again; if the file still can't be read, it runs as the demo until a later check reads it.

The plugin never contacts our servers on its own. `Check for Update` opens our website in your browser, and a bug report drafts an email in your own mail client. Nothing is sent until you press send.

IN SHORT

- **Frame it**: Sync locks the time window to the beat. Free-run spans a length of time, or a set number of cycles of a chosen frequency or note. MIDI locks to the note you play. Two cycles are easier to read than one. The divider pill sets the split between scope and spectrum, or gives the whole display to one of them.
- **Analyse it**: the spectrum (with Multi FFT for the low end) shows where the energy is, the meters show your levels, and Freeze, zoom and measurement let you study what you find. Band filters isolate the range you care about.
- **Get a second signal in**: sidechain, or link another DeltaScope by track name. Align them (ADC measures the latency; a small manual offset finishes the alignment) and the two are compared sample for sample. Correlation shows where they cancel.
- **The delta**: subtract the original from the processed signal and only the change is left. Identical signals draw a flat line; once the two are aligned, anything else is what your processor is doing: in time on the waveform, in frequency on the spectrum's Difference view, and as a response curve in Magnitude view.
- **Hear it**: Hear the Delta makes the difference signal the plugin's output, but only while Delta mode is selected; Hear the Sum does the same for Sum mode. They're the only times a licensed DeltaScope changes your audio.

LICENCE AND WARRANTY

DeltaScope is licensed, not sold. Use of the software is governed by the End User Licence Agreement shown by `Licence Agreement` on the General page of Settings and shown by the macOS installer.

Apollo View Modular Ltd supplies this software “as is” and, to the maximum extent permitted by law, accepts no responsibility for harm to persons, apparatus or data caused through the operation of this product. This does not affect your statutory rights.

Please contact support@apolloviewmodular.com with any questions, bug reports or requests.

https://www.apolloviewmodular.com