PI BY SAHARTY

PI SYSTEM ALIGN

Measure and align your sound system



USER MANUAL

Version 1.1.0 · macOS & Windows

www.pi-ware.com · support@pi-ware.com

Precision is a constant.

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WELCOME

Pi System Align by Saharty is a dual-channel measurement and alignment application for sound systems, for macOS and Windows. It measures your PA the way a system engineer does: it compares the signal you send to the system (the reference) with what a measurement microphone hears in the room. Then it tells you what to change. It shows you **where to put the microphone**, **what delay to set**, **whether to flip polarity** and **when an all-pass filter will help**, so your loudspeakers add up instead of fighting each other.

Everything you need to tune a system is in one window:

- **Transfer function**: multi-time-window dual-channel FFT with magnitude, phase, coherence and a live impulse, plus averaging, smoothing, coherence blanking, delay tracking and target curves.
- **Spectrum**: a real-time analyser from 1/1 to 1/48 octave with A, C or Z weighting, peak hold, bars and a spectrograph.
- **Impulse response**: log-sweep or live-signal measurement with EDT, T20, T30, C50, C80, D50 and Ts per octave, plus WAV import and export.
- **SPL meter**: A, C and Z weighted Fast, Slow and Leq readings, maxima, peaks, 1, 10 and 15 minute Leq values and a per-second log.
- **Alignment**: a mic placement assistant, crossover alignment, an arrival-time planner and a delay calculator with end-fire and gradient (cardioid) sub tools.
- **Demo system**: a virtual PA with a main system, a sub and a delay tower in a room, measured by two virtual microphones. You can learn every feature without an audio interface.

Who it is for. Pi System Align is for live sound engineers, system techs, installers and studio engineers who want measurements they can trust and clear advice they can act on. You don't need any measurement experience: the built-in demo system and the placement assistant walk you through the method step by step.

Version 1.1.0. This manual covers version 1.1.0, the current release. It includes all of the tools listed above, the built-in demo system and Pi by Saharty licensing, with online and offline activation and a 15-day free trial.



Figure 1. The built-in demo system: a full transfer function measurement without any hardware

QUICK START

Follow these steps to get from installation to your first aligned measurement. Steps 3 to 7 use the built-in demo system, so you don't need an interface yet.

1. **Install** Pi System Align from the `.dmg` (Mac) or the `.exe` installer (Windows). See *Installation*.
2. **Start the app.** On first launch the licence window opens. Press **Start 15-day free trial**, or enter your licence key and press **Activate**.
3. On first launch the **Demo system (built-in)** starts by itself. If audio is stopped, press **Start audio** in the command bar.
4. Press **G** (or click **Generator off**) to turn on the pink noise generator.
5. Press **D** to find the delay of the active measurement (**Mic A**). The status bar shows the delay that was found and how far the arrival stands above the noise.
6. Look at the **Transfer function** tab: the magnitude, phase and coherence traces settle as the averages build up. Press **Space** to capture a trace.
7. Open **Alignment > Crossover alignment**, choose two sources and press **Analyse**. Then press **Apply to demo system** and measure again to hear and see the improvement.
8. When you're ready for a real system, open **Audio settings**, choose your interface, wire a loopback reference and connect your measurement mic. See *Connecting a real system*.

> **Tip:** Press **Ctrl+1** to **Ctrl+5** (**Cmd** on a Mac) to jump between the five views.

INSTALLATION

macOS

1. Download the installer for your Mac from your account at pi-ware.com. There is one `.dmg` for Apple silicon and one for Intel Macs.
2. Open the `.dmg` and drag **Pi System Align** to the **Applications** folder.
3. Start the app from Applications. The app is signed with the Saharty Developer ID and notarized by Apple, so it opens like any other app.
4. When macOS asks for microphone access, choose **Allow**. macOS grants audio input this way, so without it the app receives no input signal.

> **Note:** If you declined microphone access by mistake, open **System Settings > Privacy & Security > Microphone** and switch on **Pi System Align**.

Windows

1. Download the Windows installer (`.exe`) from your account at pi-ware.com.
2. Run the installer and follow the steps. It shows the licence agreement first.
3. If Windows SmartScreen appears, choose **More info**, then **Run anyway**.
4. Start Pi System Align from the Start menu.

On Windows, Pi System Align enables ASIO automatically. ASIO drivers are listed next to the WASAPI devices in **Audio settings**.

First launch

When you start Pi System Align for the first time:

- The licence window opens once, so you can activate or start the free trial.
- Audio starts on the **Demo system (built-in)**, and the status bar says: *"The demo system is running. Turn on the generator (G) to start measuring."*
- The **Alignment > Mic placement** page already holds a typical show (mains, subs, front fills, out fills and delays) sized for a 30 m by 24 m room.

Pi System Align remembers its settings automatically: your audio device, channels, measurements, view settings, window layout and placement plan are restored the next time you open it.

ACTIVATION & LICENSING

Pi System Align uses the Pi by Saharty licensing system. A licence is a one-time purchase with free updates.

Getting a licence

Buy Pi System Align at **pi-ware.com/products/pi-system-align**. Your licence key arrives by email after purchase. It looks like `XXXXX-XXXXX-XXXXX-XXXXX`.

The licence window

Open the licence window from **Help > Licence...**, or click the coloured banner at the top of the main window. It contains:

Control	What it does
Licence key	Paste or type your licence key here.
Activate	Activates this computer online with the key you entered.
Start 15-day free trial	Starts the full-featured free trial. You don't need any payment details.
Deactivate this computer	Frees this computer so the licence can be activated on another one.
Offline activation	Shows the request code for activating a computer that has no internet connection.
Copy code	Copies the offline request code.
Load licence file...	Loads the licence file (`.pilic`) you got from the website.
Buy a licence	Opens the product page.

Online activation

1. Open **Help > Licence...**.
2. Paste your key into **Licence key** and press **Activate** (or press Enter).
3. The status line reads *"Activated ... Every feature is on."*

Free trial

Press **Start 15-day free trial**. Every feature works during the trial, and the banner at the top of the window shows the days that are left.

Offline activation

If the measurement computer never goes online:

1. Enter your licence key in the licence window.

2. Click **Offline activation**. A request code appears.
3. Press **Copy code**, and move the code to any device that has internet access, for example on a USB stick or in a message to yourself.
4. Paste the code at **pi-ware.com/activate**. You receive a licence file.
5. Bring the file back and press **Load licence file...**.

Moving to a new computer

One licence activates one computer.

- **From the old computer:** open the licence window and press **Deactivate this computer**, then activate on the new one.
- **If the old computer is not available:** activate on the new computer anyway. Pi System Align tells you the key is active on another computer and offers **Move it here**. The other computer gets a notice.
- **From your account:** you can deactivate computers or reset machines from your account at **pi-ware.com/account**.

How the licence check works

- The app checks the licence whenever it finds an internet connection: at launch, then every hour, and every 2 minutes while offline. It keeps working offline with no time limit.
- **24-hour rule:** nothing is ever limited without at least 24 hours of notice on that computer. This applies when a licence is moved, a trial ends, a refund is made or a computer is deactivated from the account. During the notice every feature works, and an amber banner shows the message and the countdown.

Limited mode

If the app has never been activated, or the notice has run out, Pi System Align runs in limited mode. A red banner explains the situation. Everything can still be opened, measured and viewed: audio, the generator, the transfer function, spectrum, impulse response, SPL meter, alignment tools, sessions and imports. Saving and exporting are turned off. This covers saving sessions, trace CSV export, impulse response WAV export, the SPL log and the mic placement plan. These buttons and menu items read **Activate to save** and open the licence window.

> **Note:** Pi System Align never quits and never stops a running measurement because of the licence. Your show is safe.

The licence file is stored here:

- **Mac:** `~/Library/Application Support/Pi by Saharty/Licences/pi-system-align.pilic`
- **Windows:** `%APPDATA%\Pi by Saharty\Licences\pi-system-align.pilic`

INTERFACE TOUR

The main window has five areas: the **licence banner** (shown only when needed), the **command bar**, the **sidebar** on the left, the **analysis tabs** in the centre and the **status bar** at the bottom.

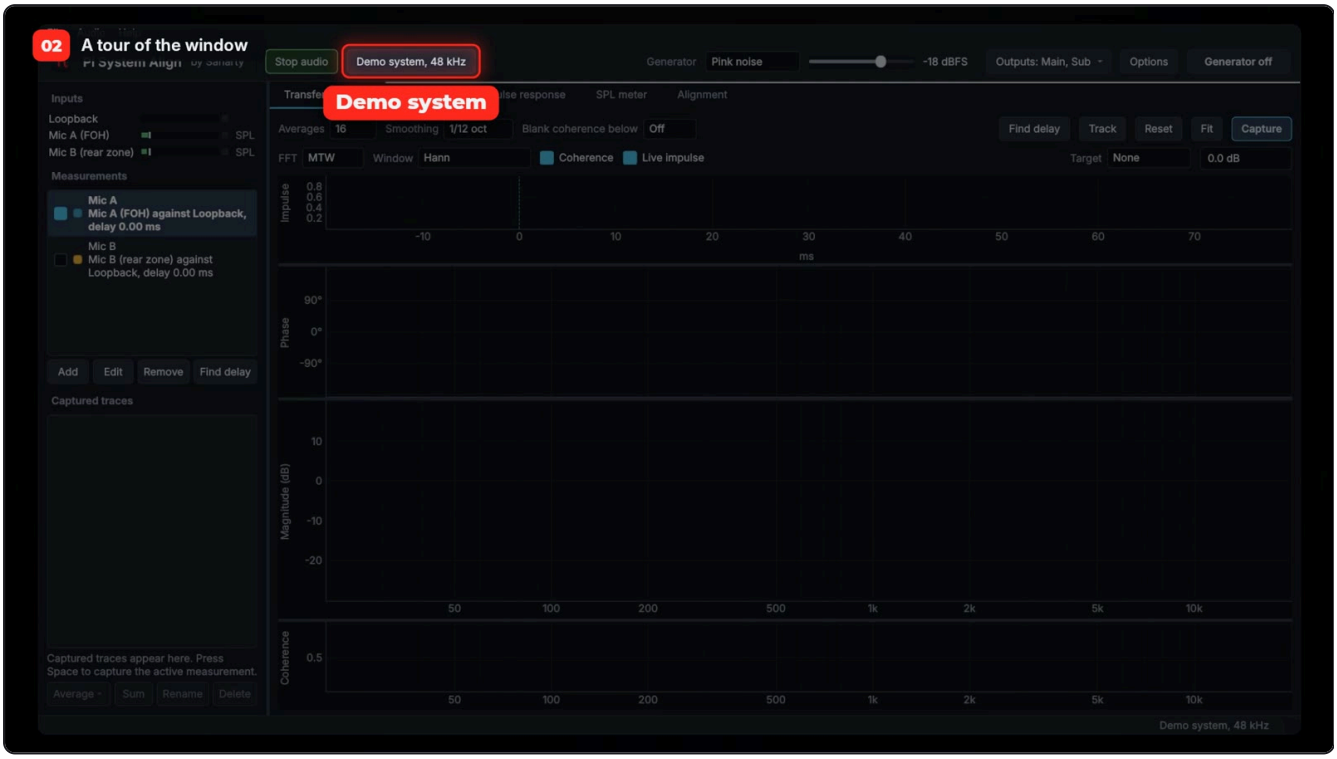


Figure 2. The main window: command bar, sidebar with inputs, measurements and traces, and the analysis tabs

Command bar

The command bar runs across the top. It holds the audio controls and the signal generator.

Control	What it does	Range/default
Start audio / Stop audio	Starts or stops audio input and output.	—
Audio settings	Opens the audio settings. While running, the button shows the device and sample rate in use.	—
Generator type	Signal type: **Pink noise**, **Pink periodic**, **White noise**, **Sine** or **Log sweep**.	Pink noise
Frequency (Sine only)	Sine frequency.	10–24000 Hz, default 1000 Hz
Level slider	Generator level in dBFS (sine reference).	–60 to 0 dBFS, default –18 dBFS
Outputs	Chooses which outputs the generator feeds. It includes **All outputs** and **No outputs**.	—

Control	What it does	Range/default
Options	Opens **Generator options**: band limits, sine frequency and sweep length.	—
Generator off / Generator on	Turns the generator on or off (**G**).	Off

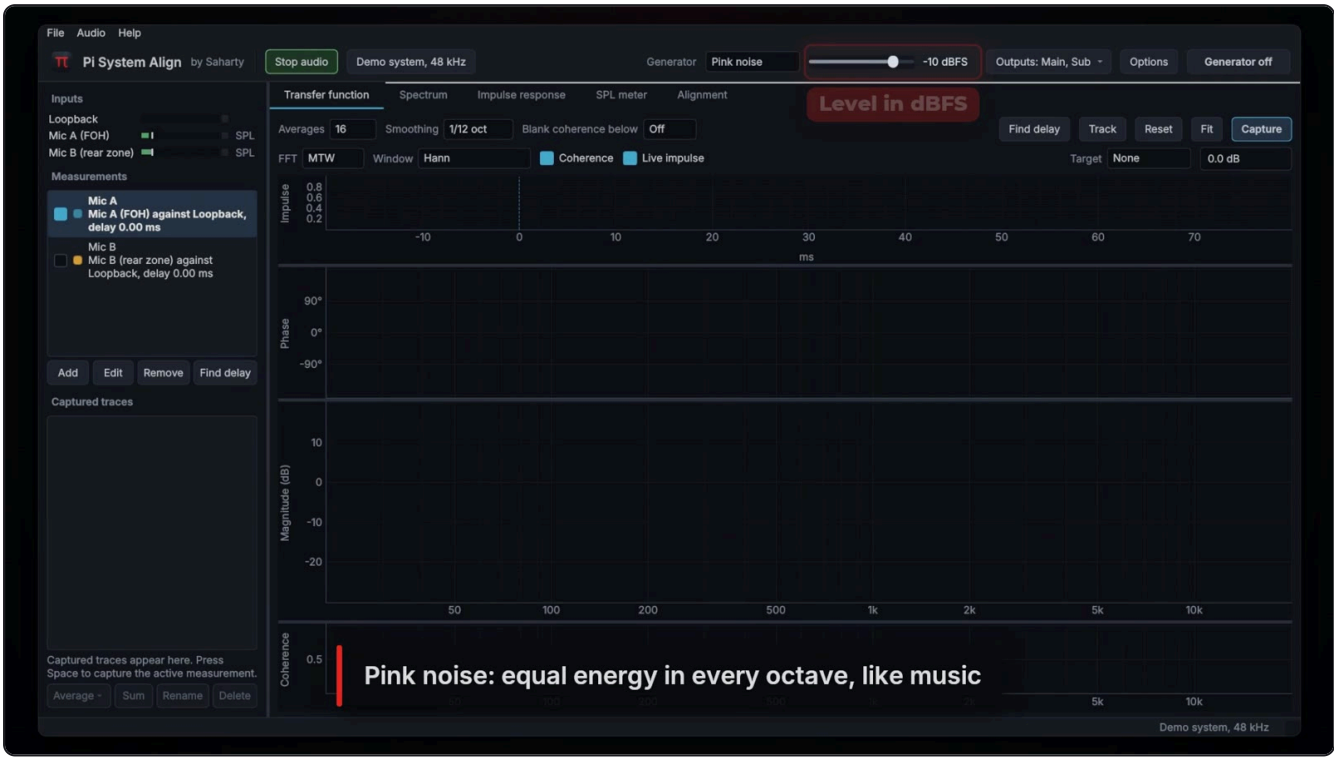


Figure 3. The generator controls in the command bar: type, level, outputs and the on/off button

Generator options contains:

Control	What it does	Range/default
Limit the noise to a band	Band-limits the noise between **Low** and **High**.	Off
Low	Lower band limit.	5–20000 Hz, default 20 Hz
High	Upper band limit.	10–24000 Hz, default 20000 Hz
Sine frequency	Frequency of the sine.	10–24000 Hz
Sweep length	Length of the log sweep.	1.4 s, 2.7 s, 5.5 s, 10.9 s; default 2.7 s

> **Warning:** The generator can play loud signals. Always start with the level low and the amplifiers turned down, and bring the level up gradually. Protect your hearing, your audience and your loudspeakers.

Sidebar

The sidebar has three sections.

Inputs lists every input channel with a peak meter. A calibrated input shows **SPL** next to its meter. Right-click an input name for **Input settings...** and **Calibrate...**.

Measurements lists your measurements. A measurement is a pair: a microphone input measured against a reference. Each entry shows its name, the inputs used, the delay compensation and whether delay tracking is on.

Control	What it does
Tick box	Runs (ticked) or pauses the measurement.
Selecting an entry	Makes it the **active** measurement (shown in bold). Capture, Find delay and the live impulse act on the active measurement.
Add	Adds a measurement.
Edit	Edits the selected measurement (or double-click it).
Remove	Removes the selected measurement.
Find delay	Finds the delay of the selected measurement (**D**).
Right-click menu	**Add measurement...**, **Edit...**, **Find delay**, **Track delay**, **Remove**.

Captured traces lists the traces you have captured or imported. Tick a trace to show it and untick it to hide it. Select several traces with Shift or Ctrl/Cmd.

Control	What it does
Average	Averages the selected traces: **Power average**, **Complex average** or **dB average**.
Sum	Complex sum of the selected traces. It predicts how the sources sound when they play together.
Rename	Renames the selected trace.
Delete	Deletes the selected traces.
Right-click menu	**Edit offset, delay and polarity...**, **Colour...**, **Rename...**, **Export as CSV...**, **Show only selected**, **Show all**, **Hide all**, **Import transfer function...**, **Import spectrum...**, **Delete**.

Analysis tabs

Tab	Shortcut	Purpose
Transfer function	Ctrl/Cmd+1	Live impulse, phase, magnitude and coherence of your measurements.
Spectrum	Ctrl/Cmd+2	Real-time analyser and spectrograph.
Impulse response	Ctrl/Cmd+3	Sweep measurement, decay curves and room acoustics.
SPL meter	Ctrl/Cmd+4	Weighted sound level meter with a log.
Alignment	Ctrl/Cmd+5	**Mic placement**, **Crossover alignment**, **Arrival times**, **Delay calculator**.

Status bar

The left side of the status bar shows messages, for example the result of **Find delay**. On the right you see the cursor readout (frequency, level, phase and coherence under the mouse), **Dropouts** (audio buffer under-runs, shown only when there are any) and the audio engine information.

Menus

Menu	Items
File	**Open session...**, **Save session...**, **Import transfer function...**, **Import spectrum...**, **Quit**
Audio	**Audio settings...**, **Start or stop audio**, **Demo system processor...**, **Reset averages**
Help	**Keyboard shortcuts**, **Licence...**, **About Pi System Align by Saharty**

HOW DUAL-CHANNEL MEASUREMENT WORKS

Pi System Align compares two signals. The **reference** is the signal you send to the system. The **measurement** is what the microphone hears after the processor, amplifiers, loudspeakers and room. The difference between them is the **transfer function**: what the speakers and the room did to your signal. Because the analysis compares the two signals, it works with pink noise and also with any broadband programme, such as music or speech.

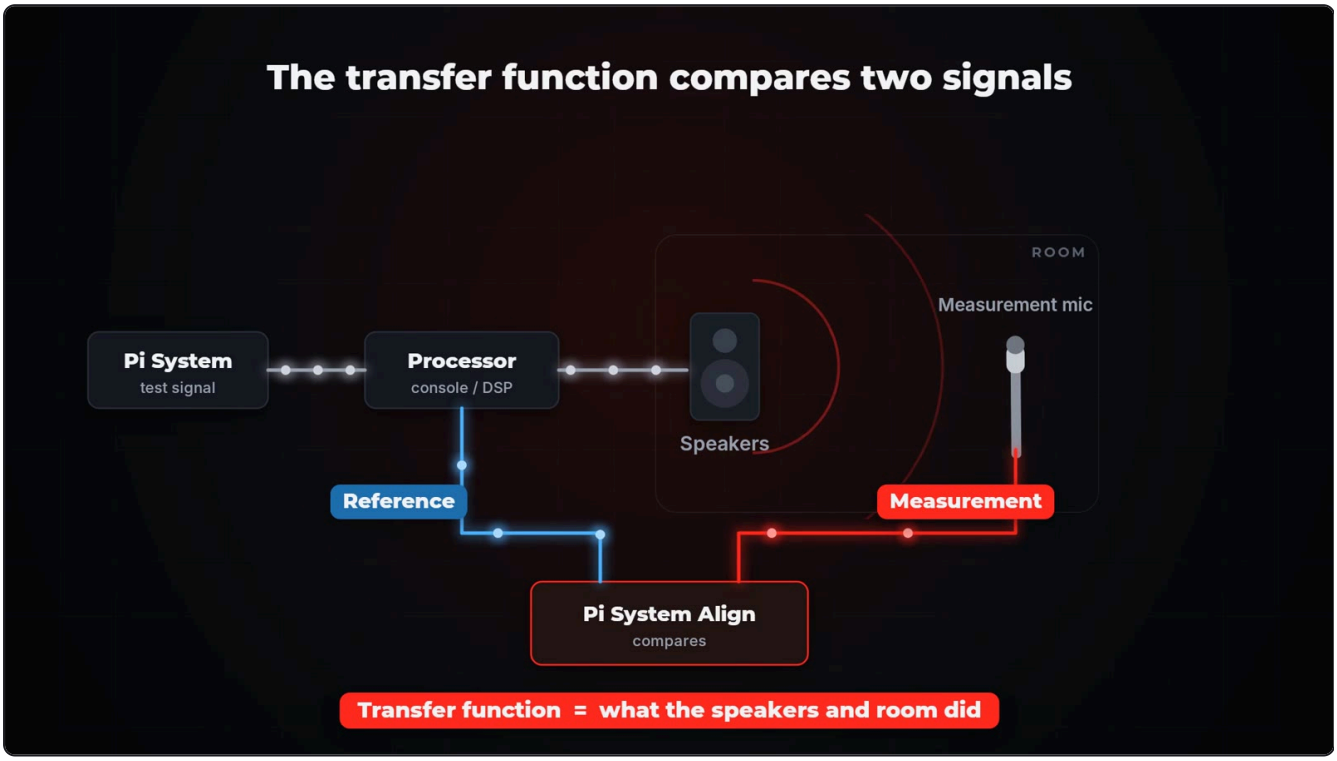


Figure 4. Dual-channel measurement: the reference against the measurement mic gives the transfer function

When two loudspeakers cover the same seat, their sound adds up there. If one arrives a little late, some frequencies add while others cancel. Even 1 ms of difference creates deep cancellations across the spectrum. Alignment means changing **delay**, **polarity** and sometimes the phase (with an **all-pass** filter), so the sources arrive together where they overlap.

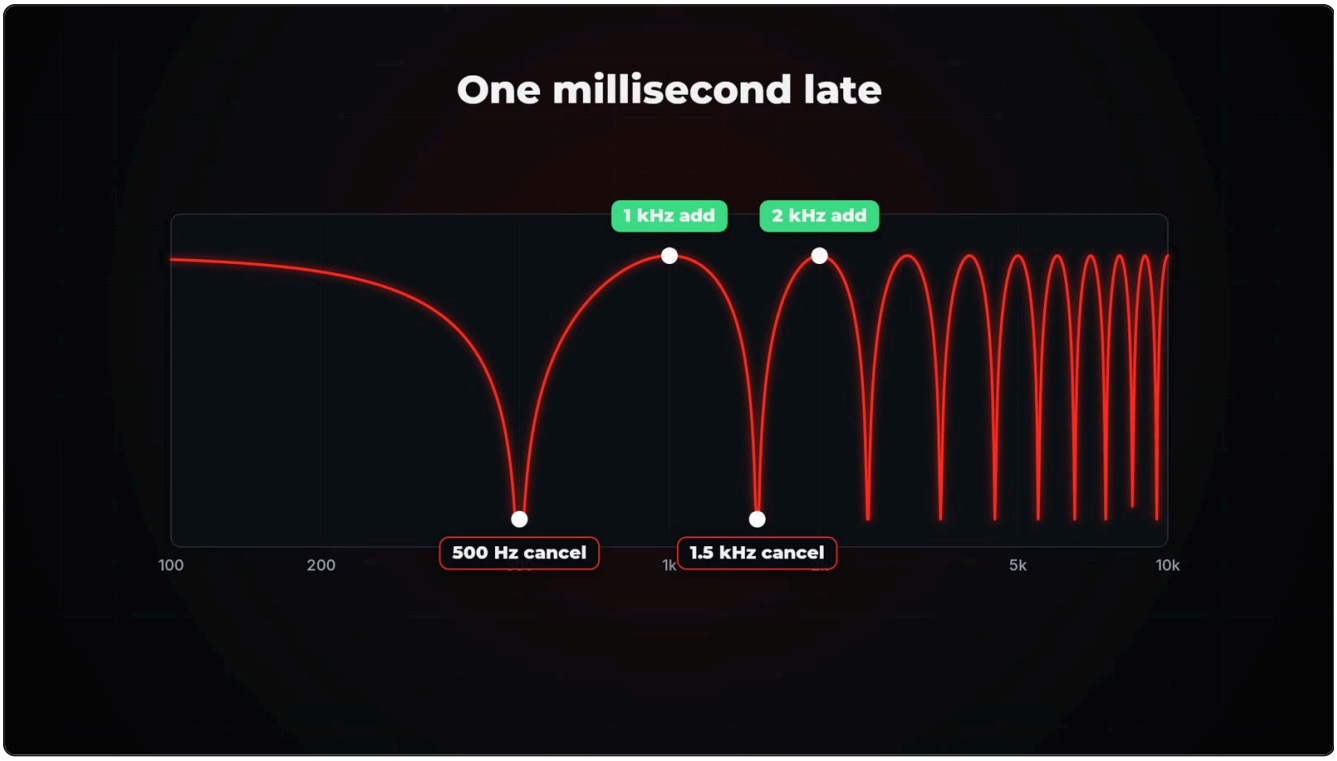


Figure 5. One millisecond late: the sum of two sources forms a comb of cancellations



Figure 6. The three alignment tools: delay, polarity and all-pass

PRACTISING WITH THE DEMO SYSTEM

The **Demo system (built-in)** is a virtual PA in a room. It has three outputs (**Main**, **Sub**, **Delay tower**) and three inputs (**Loopback**, **Mic A (FOH)**, **Mic B (rear zone)**). The virtual microphones are pre-calibrated, so they read in dB SPL. Two measurements are set up for you: **Mic A** (active) and **Mic B** (paused).

How to take your first demo measurement

1. Make sure **Demo system (built-in)** is selected in **Audio settings**, then press **Start audio**.
2. Under **Outputs**, choose which virtual speakers play: for example **Main** only, **Sub** only, or both.
3. Press **G** to start the pink noise.
4. Press **D** to find the delay of **Mic A**.
5. Watch the **Transfer function** tab fill in, then press **Space** to capture.



Figure 7. Pink noise on the demo system: the transfer function settles as averages build up ▶ Animated in the online manual

The demo system processor

Open **Audio > Demo system processor...** (or **Demo processor...** on the **Crossover alignment** page). It works exactly like a real loudspeaker processor, one row per output:

Control	What it does	Range/default
Delay	Output delay.	0–2000 ms, default 0 ms
Polarity	**Invert** flips the polarity.	Off

Control	What it does	Range/default
Gain	Output gain.	–40 to +12 dB, default 0 dB
Mute	Mutes the output.	Off
All-pass	**Off**, **1st order** or **2nd order**.	Off
Frequency	All-pass frequency.	10–2000 Hz, default 100 Hz
Q	All-pass Q (2nd order).	0.3–10, default 0.71
Reset all	Returns every output to its default.	—

Changes apply immediately. Use the processor to practise each alignment tool, then measure again to check the result.

CONNECTING A REAL SYSTEM

How to choose your interface

1. Click **Audio settings** in the command bar.
2. Choose the **Input device** and the **Output device**. If the input device also has outputs, it is selected for output automatically. Use the same interface for input and output for stable timing.
3. Choose the **Sample rate** and **Block size**.
4. Set the number of **Input channels** and **Output channels**.
5. Press **Apply and start**.

Control	What it does	Range/default
Input device	Interface for the reference and microphone inputs. Includes **Demo system (built-in)**.	—
Output device	Interface the generator plays through, or **No output**.	—
Sample rate	Audio sample rate.	44100, 48000, 88200, 96000 Hz; default 48000 Hz
Block size	Audio buffer size in samples.	256, 512, 1024, 2048, 4096; default 1024
Input channels	Number of inputs to open.	Up to 64 (the device's maximum)
Output channels	Number of outputs to open.	Up to 64 (the device's maximum)
Rescan devices	Looks for interfaces again.	—



Figure 8. Audio settings: choose the interface, sample rate and channels (48 kHz is usual)

The reference: loopback or internal generator

For the best timing, use a **loopback**. Take the signal you send to the system (for example a spare output of your interface, or a matrix output from the console or processor) and wire it back into an input. That input becomes the reference.

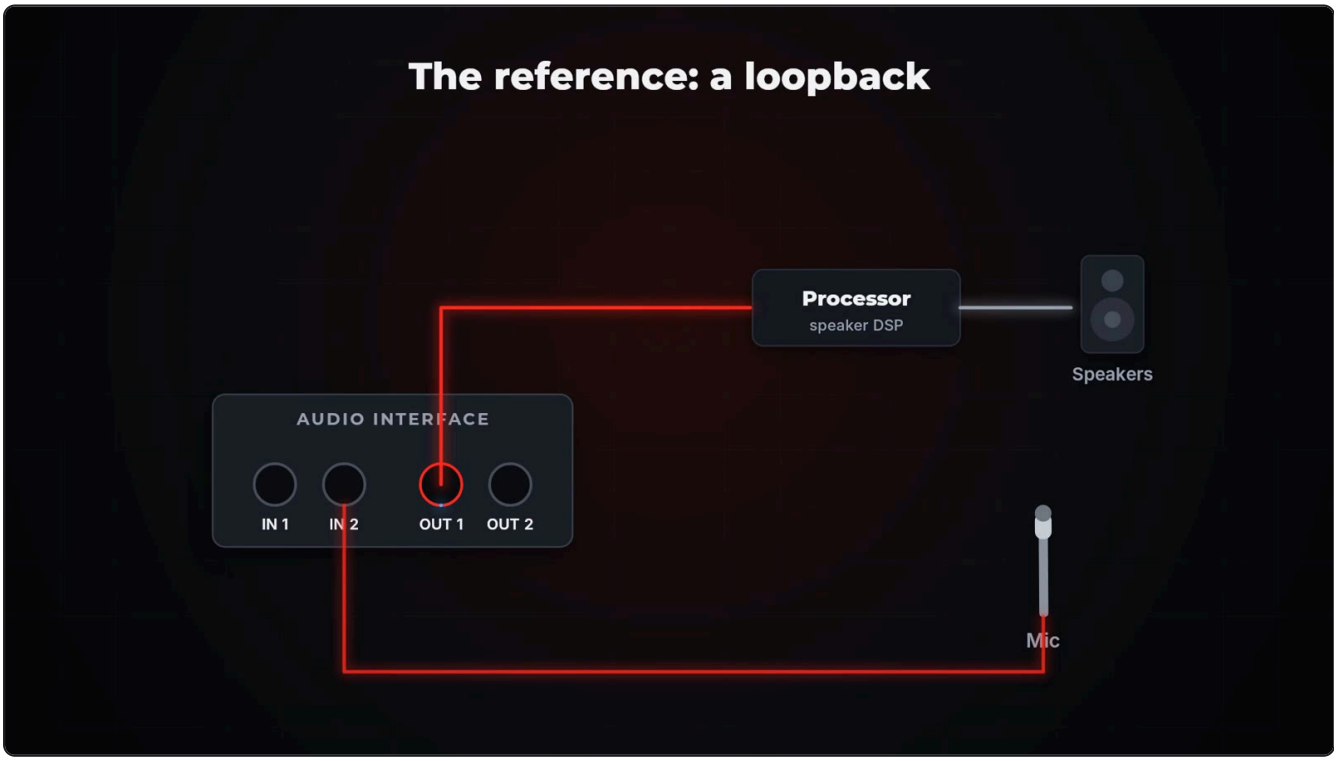


Figure 9. Loopback reference: the signal sent to the system is wired back into an input

If you can't wire a loopback, choose **Generator (internal reference)** as the reference. This works, but a loopback gives the most reliable timing because it includes everything in your own signal path.

The measurement microphone

Use an omnidirectional measurement microphone, with +48 V phantom power if it needs it. Set the preamp gain so the input meter sits comfortably in the green and amber with pink noise playing, and never in the red.

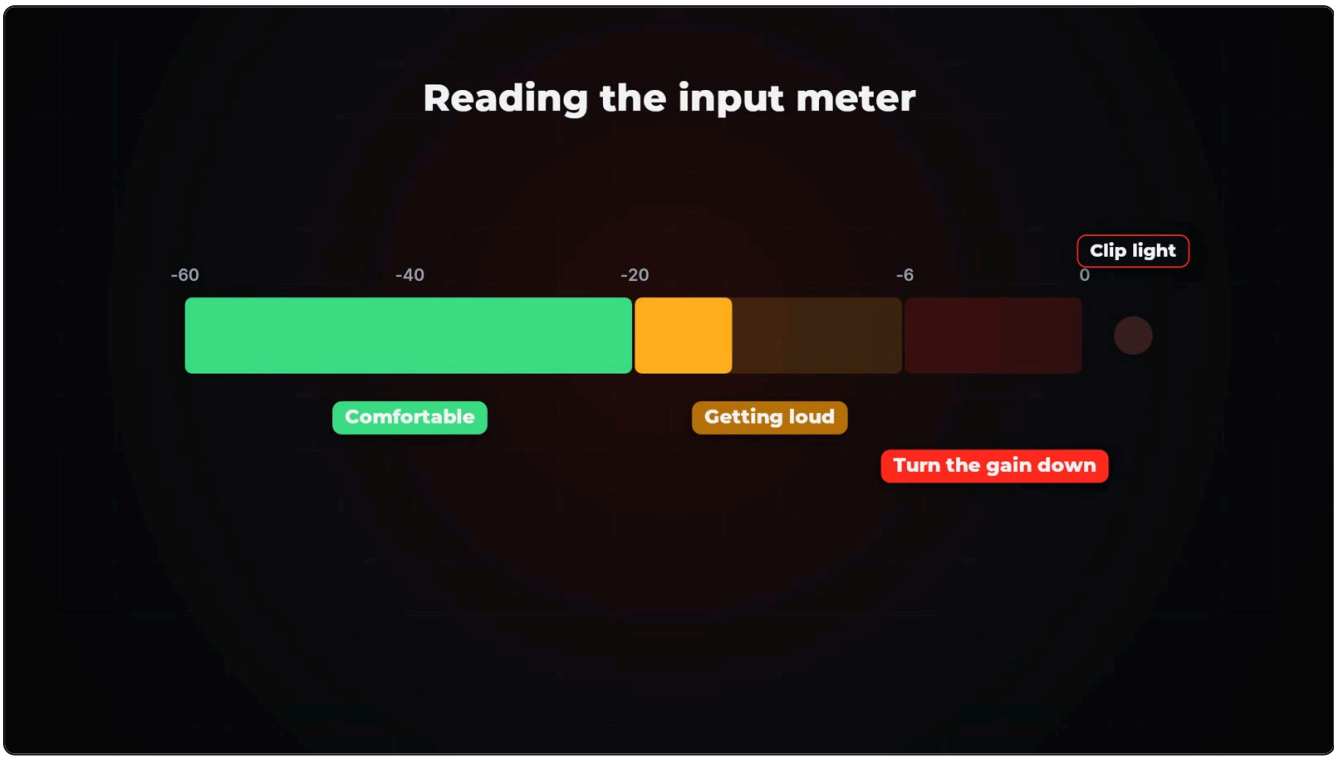


Figure 10. Reading the input meter: green is comfortable, amber is getting loud, red means turn the gain down

INPUTS, CALIBRATION AND MEASUREMENTS

How to set up an input

1. Right-click the input name in the sidebar and choose **Input settings...**.
2. Give it a clear **Name**, for example *Loopback* or *Mic FOH*.
3. To remove your microphone's own response, press **Load...** under **Mic correction** and choose its correction file. This is a text or CSV file with frequency and dB columns (`.txt` , `.csv` , `.frd` or `.cal`). Press **Clear** to remove it.
4. Press **OK**.

Control	What it does	Range/default
Name	Input name shown everywhere in the app.	—
Calibrated: 0 dBFS equals	Turns calibration on and sets the SPL that full scale represents.	0–200 dB SPL
Calibrate...	Measures an acoustic calibrator to set the level.	—
Mic correction	**Load...** or **Clear** a microphone correction curve.	None

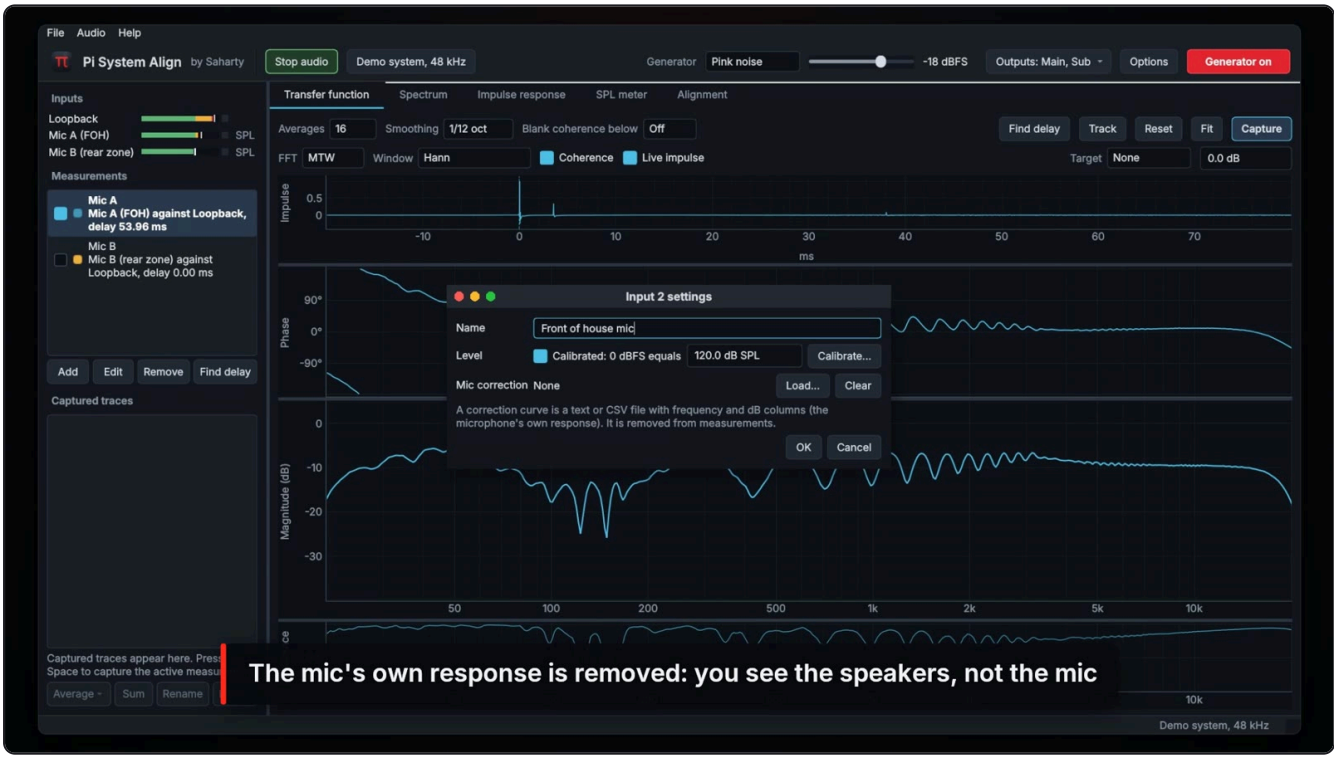


Figure 11. Input settings: name, level calibration and microphone correction

How to calibrate a microphone input

Calibration makes the analyser and the SPL meter read in dB SPL instead of dBFS.

1. Start audio.

2. Fit the acoustic calibrator on the microphone and switch it on.
3. Right-click the input and choose **Calibrate…** (or press **Calibrate…** on the **SPL meter** tab).
4. Choose the **Calibrator level (dB SPL)**: **94.0**, **104.0** or **114.0**, or type your calibrator's level.
5. Wait a few seconds, then press **Measure**. The dialog shows the frequency and level it measured, and what full scale will read.
6. Press **Use this calibration**.

> **Note:** If the strongest tone is not close to 1 kHz, Pi System Align warns you to check the calibrator. If the input is within 3 dB of clipping, it asks you to lower the preamp gain and calibrate again.

How to add a measurement

1. In **Measurements**, press **Add**.
2. Enter a **Name**, for example *FOH centre*.
3. Choose the **Measurement input** (the microphone) and the **Reference** (the loopback input, or **Generator (internal reference)**).
4. Leave **Delay compensation** at 0 ms. **Find delay** fills it in for you.
5. Pick a **Colour** and press **OK**.

Control	What it does	Range/default
Name	Measurement name.	Mic 1, Mic 2, …
Measurement input	The microphone channel.	—
Reference	Loopback input or internal generator.	—
Delay compensation	Time offset between reference and microphone.	0–2000 ms, default 0 ms
Colour	Trace colour.	Next free colour

You can run several measurements at once, for example two microphones against the same loopback. Tick each one you want to run, and select the one you want to work on.



Figure 12. Two microphones measured at once against the same reference ▶ **Animated in the online manual**

MEASURING THE TRANSFER FUNCTION

The **Transfer function** tab shows four stacked plots: **Impulse** (the live impulse response), **Phase**, **Magnitude (dB)** and **Coherence**. Phase, magnitude and coherence share the same frequency axis.



Figure 13. The transfer function view: impulse, phase, magnitude and coherence

Toolbar controls

Control	What it does	Range/default
Averages	Number of averages.	1, 2, 4, 8, 16, 32, 64, 128, Inf; default 16
Smoothing	Frequency smoothing of the display.	None, 1/48, 1/24, 1/12, 1/6, 1/3, 1 oct; default 1/12 oct
Blank coherence below	Hides magnitude and phase where coherence is below the threshold.	Off, 10%–50%; default Off
FFT	**MTW** (multi-time-window) or a fixed FFT size.	MTW, 4096, 8192, 16384, 32768, 65536; default MTW
Window	FFT window.	Hann, Blackman-Harris, Hamming, Flat top, Rectangular; default Hann
Coherence	Shows or hides the coherence plot.	On
Live impulse	Shows or hides the impulse plot.	On
Target	Draws a target curve on the magnitude plot.	None, Flat, Tilt, Live sound, Custom...; default None

Control	What it does	Range/default
Target offset	Moves the target curve up or down.	−80 to +80 dB, default 0 dB
Find delay	Finds the delay of the active measurement (**D**).	—
Track	Keeps following the delay of the active measurement.	Off
Reset	Restarts averaging (**R**).	—
Fit	Fits the magnitude scale to the active measurement.	—
Capture	Captures the active measurement as a trace (**Space**).	—

MTW uses long FFTs at low frequencies and short ones at high frequencies. This gives you fine resolution in the low end and a fast, clean response in the highs. The fixed sizes are useful when you want the same resolution at every frequency.

Target curves: Flat is a 0 dB line. **Tilt** slopes at −3 dB per decade around 1 kHz. **Live sound** adds a low-end shelf of about +7 dB and rolls off about 1 dB per octave above 2 kHz. **Custom…** loads your own curve from a `.txt`, `.csv` or `.frd` file with frequency and dB columns. When a target is shown, **Fit** also moves the target to the level of your measurement.

How to find the delay

1. Turn on the generator (**G**) with the right outputs selected.
2. Press **Find delay** (or **D**).
3. The status bar reports the delay, for example *"Mic A: delay 42.17 ms, peak 38 dB above the noise."* If the main impulse peak is negative, the message also says *"The impulse peak is negative."* This hints at a polarity inversion somewhere in the path.

The delay is stored in the measurement's **Delay compensation**, so the phase trace becomes readable and the coherence rises. The delay finder searches up to one second.

> **Tip:** Turn on **Track** (or **Track delay** in the measurement's right-click menu) when the delay can drift, for example outdoors with wind and changing temperature. Pi System Align checks the delay again every 2 seconds.

How to read the plots

- **Magnitude** shows the level at each frequency. Peaks and dips are where you might use EQ, or where sources interfere.
- **Phase** shows the timing at each frequency. A flat phase in the crossover region means the sources are in time there.

- **Coherence** (0 to 1) tells you how much to trust the data. Above about 0.9 is good. Low coherence means noise, reflections or a position in a deep cancellation.
- **Impulse** shows the arrival in time. A clean, single peak at 0 ms means the delay is right.

Move the mouse over a plot to read the exact frequency, level, phase and coherence in the status bar.



Figure 14. Hover over a plot to read frequency, level, phase and coherence in the status bar ▶ Animated in the online manual

WORKING WITH CAPTURED TRACES

A **captured trace** is a snapshot of a measurement. It keeps the magnitude, phase and coherence, plus the delay compensation that was in use when you captured it.

How to capture, compare and edit traces

1. Press **Space** (or **Capture**) to capture the active measurement. On the **Spectrum** tab, **Space** captures the spectrum of each shown input.
2. Tick or untick traces to compare them with the live measurement.
3. Double-click a trace (or right-click > **Edit offset, delay and polarity...**) to change its **Name**, **Colour**, **Level offset** (–60 to +60 dB), **Added delay** (–1000 to +1000 ms) and **Invert polarity**. The dialog also shows the delay compensation at capture time.



Figure 15. A captured trace keeps its delay, so you can compare it with later measurements

How to average positions

One microphone position doesn't tell you the whole story. Measure several seats, capture each one, then average them, and make your EQ decisions on the average.

1. Capture a trace at each position.
2. Select the traces (Shift-click or Ctrl/Cmd-click).
3. Press **Average** and choose **Power average**, **Complex average** or **dB average**.

Power average is the usual choice for averaging several positions. **Complex average** keeps the phase, and is only meaningful when the traces share the same timing.



Figure 16. Walk several positions, then make EQ decisions on the average

How to predict a combined response

Select two or more transfer function traces and press **Sum**. Pi System Align calculates the complex sum, which is the response you would get with those sources playing together, including their delay and polarity. Every trace in the sum needs phase data.



Figure 17. Mains plus sub: the complex sum predicts the combined response

Import and export

- **Import:** choose **File > Import transfer function...** or **Import spectrum...** (or use the traces right-click menu) to load a `.csv`, `.txt` or `.frd` file. For a transfer function the columns are frequency, magnitude (dB) and optionally phase (degrees) and coherence. For a spectrum they are frequency and level.
- **Export:** right-click a trace and choose **Export as CSV...**. Transfer functions are exported with frequency, magnitude, phase and coherence. Spectra are exported with frequency and level.

USING THE SPECTRUM ANALYSER

The **Spectrum** tab shows the energy at each frequency for one or more inputs. Unlike the transfer function, it doesn't need a reference.

Control	What it does	Range/default
Banding	Frequency resolution.	1/1, 1/3, 1/6, 1/12, 1/24, 1/48 oct; default 1/3 oct
Averaging	Time averaging.	None, 0.25 s, 0.5 s, 1 s, 2 s, 4 s, 8 s, Inf; default 1 s
Weighting	Frequency weighting.	Z, A, C; default Z
Inputs	Chooses which inputs the analyser shows.	—
Peak hold	Draws dashed peak-hold traces.	Off
Bars	Draws octave and third-octave bands as bars (first input shown).	On
Spectrograph	Shows the spectrograph under the analyser.	Off
Range	Spectrograph colour range.	40, 60, 80, 100 dB; default 60 dB
Clear peaks	Clears the peak-hold traces.	—
Reset	Restarts averaging (**R**).	—
Capture	Captures the spectrum of each shown input (**Space**).	—

Levels read in **dB SPL** when the input is calibrated and in **dBFS** when it is not. Bars appear only at 1/1 and 1/3 octave banding.



Figure 18. Third-octave bars on the spectrum analyser ▶ Animated in the online manual

The **Spectrograph** shows the last ten seconds as a scrolling colour map. A frequency that stays bright in the same place, like a monitor that is about to feed back, shows up as a steady line.



Figure 19. The spectrograph shows the last ten seconds at a glance ▶ Animated in the online manual

MEASURING THE IMPULSE RESPONSE AND ROOM ACOUSTICS

The **Impulse response** tab measures how the room responds over time and calculates the standard room acoustics parameters.

Control	What it does	Range/default
Source	**Sweep** plays a log sweep and gives the most accurate room acoustics. **Live signal** uses the running generator or programme: quick and indicative, with less range at high frequencies.	Sweep
Sweep	Sweep length. Lengths that don't fit the audio buffer are greyed out.	1.4, 2.7, 5.5, 10.9 s; default 2.7 s
Averages	Number of sweeps averaged.	1–8, default 1
Input	The measurement microphone.	—
Reference	**Generator (internal)** or an input.	—
Measure	Starts the measurement.	—
Cancel	Stops a running measurement.	—
Import WAV	Analyses an impulse response from a WAV file.	—
Export WAV	Saves the impulse response as a WAV file.	—
Display	**Log (ETC)**, **Linear** or **Decay (Schroeder)**.	Log (ETC)

How to measure a room

1. Choose the outputs that feed the system under **Outputs** in the command bar.
2. Set **Source** to **Sweep**, choose a **Sweep** length and the number of **Averages**.
3. Choose the **Input** (your microphone) and the **Reference**.
4. Press **Measure**. Keep the room quiet while the sweep plays.
5. Read the result. The status line reports the arrival time and the impulse-to-noise ratio.



Figure 20. An impulse response with its decay curve, octave-band table and direct-sound response

The table lists the **Broadband** values and the octave bands from 63 Hz to 8 kHz:

Column	Meaning
EDT (s)	Early decay time.
T20 (s), **T30 (s)**	Reverberation time from the –5 to –25 dB and –5 to –35 dB parts of the decay.
C50 (dB)	Clarity for speech.
C80 (dB)	Clarity for music.
D50 (%)	Definition: the share of energy in the first 50 ms.
Ts (ms)	Centre time.

The **Direct sound (dB)** plot shows the frequency response of the direct sound. The dashed vertical line marks where the analysis truncates the decay.



Figure 21. Clarity values (C50 and C80) per octave band

> **Note:** A T20 or T30 value shown in amber means the decay was not a clean straight line. Hover over it to see the fit quality, and treat it with care.

USING THE SPL METER

The **SPL meter** tab is a weighted sound level meter with Leq windows, maxima, peaks and a history.

Control	What it does	Default
Input	The input to meter. Uncalibrated inputs are marked *(not calibrated)*.	–
Main reading	The large reading: LAeq, LAF, LAS, LCeq, LCF, LZF, LAeq 1 min, LAeq 10 min, LAeq 15 min, LCeq 15 min or LCpeak.	LAeq
Calibrate…	Calibrates this input with an acoustic calibrator.	–
Export log	Saves the per-second log as CSV.	–
Reset	Restarts Leq, maxima and the log.	–

The grid shows LAF, LAS, LAeq, LAFmax, LCF, LCS, LCeq, LCpeak, LZF, LZeq, LAeq 1 min, LAeq 10 min, LAeq 15 min, LCeq 1 min, LCeq 15 min and LZpeak. **Fast** uses a 1/8-second time constant and **Slow** uses 1 second. The running time shows under the main reading.



Figure 22. The SPL meter: a large main reading and a grid of weighted levels ▶ **Animated in the online manual**

The history plot shows LAeq, LCeq and LAFmax for each second, covering up to the last hour. **Export log** writes a CSV with a time stamp and the per-second LAeq, LCeq, LZeq, LAFmax, LCFmax, LCpeak and LZpeak values.

> **Warning:** The SPL meter is not a certified sound level meter (it is not an IEC 61672 class 1 or class 2 instrument). Don't use it as the only basis for noise-limit compliance or any legal or contractual measurement.

Without calibration the meter reads in dBFS. Calibrate the input first (see *Inputs, calibration and measurements*) to read in dB SPL.

PLANNING MICROPHONE POSITIONS

The **Alignment > Mic placement** page tells you where to put the microphone for each loudspeaker system, explains why, and tells you how to measure it.

How to build a placement plan

1. Under **Room**, enter **Depth, stage to far seats** (5–400 m, default 30 m) and **Width of the audience** (4–300 m, default 24 m). Choose **Mic height**: **Standing, 1.7 m** or **Seated, 1.2 m**.
2. Under **Speaker systems**, press **Typical show** for mains, subs, front fills, out fills and delays. Or use **Add system** to add **Mains**, **Subs**, **Center fill**, **Front fills**, **Out fills**, **Under balcony** or **Delays** one by one. **Remove** deletes the selected system.
3. Select a system and set its sizes under **Selected system**. You can also rename it there. Sizes you don't edit follow the room.
4. Read the plan view. Squares are speakers and numbered circles are mic positions, coloured by system.
5. Select a row in the table (**Mic**, **System**, **Position**, **Where**) to see the full instructions on the right: the reason for the position, what to align to, and numbered measuring steps.

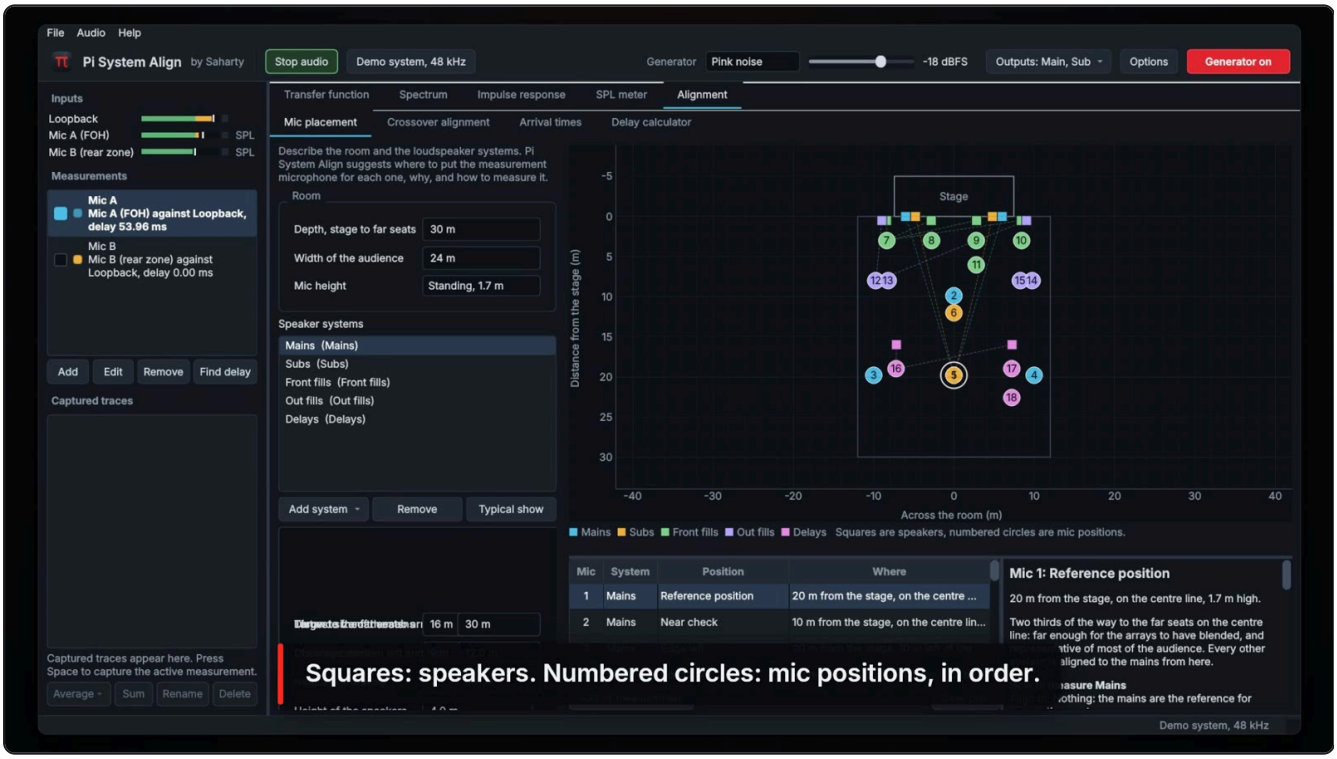


Figure 23. The placement plan: speakers, numbered mic positions and step-by-step instructions ▶ Animated in the online manual

System sizes

System	Settings (range)
Mains	Throw to the far seats (5–250 m), Distance between left and right (0–60 m), Height of the array (1–30 m)

System	Settings (range)
Subs	Largest size of the sub array (0.5–30 m)
Center fill	Coverage depth (1–40 m), Height of the speaker (0.2–10 m)
Front fills	Number of fills (1–16), Coverage depth (1–20 m), Height of the speakers (0.2–5 m)
Out fills	Throw along the side (3–100 m), Height of the speakers (0.5–20 m)
Under balcony	Overhang starts at (2–200 m), Coverage depth (2–60 m), Height of the speakers (1–10 m)
Delays	Distance from the mains (3–250 m), Coverage depth (2–100 m), Number of towers (1–12), Height of the speakers (0.5–20 m)

What the plan suggests

The plan always lists the systems in measuring order: mains first, then subs, center fill, front fills, out fills, under-balcony fills and delays.

- **Mains:** a **Reference position** two-thirds of the way to the far seats on the centre line, plus a **Near check** and **Edge left** / **Edge right** checks. Every other system is aligned to the mains from the reference position.
- **Subs:** a **Crossover position** at exactly the same spot as the mains reference, plus a **Front coverage check**.
- **Fills** (center, front, under balcony): a coverage position and a **Blend position** where the fill hands over to the mains.
- **Out fills:** a coverage position and a **Seam** position on each side, where the mains' coverage edge meets the out fill.
- **Delays:** a coverage position for each tower and a **Blend position** further into the zone.

For fills and delays, the notes give a **starting delay** calculated from the geometry and the air temperature. They also give the value with an 8 ms image offset, which keeps the sound image on the stage. Warnings appear in amber, for example when the room is small for the system sizes you entered.

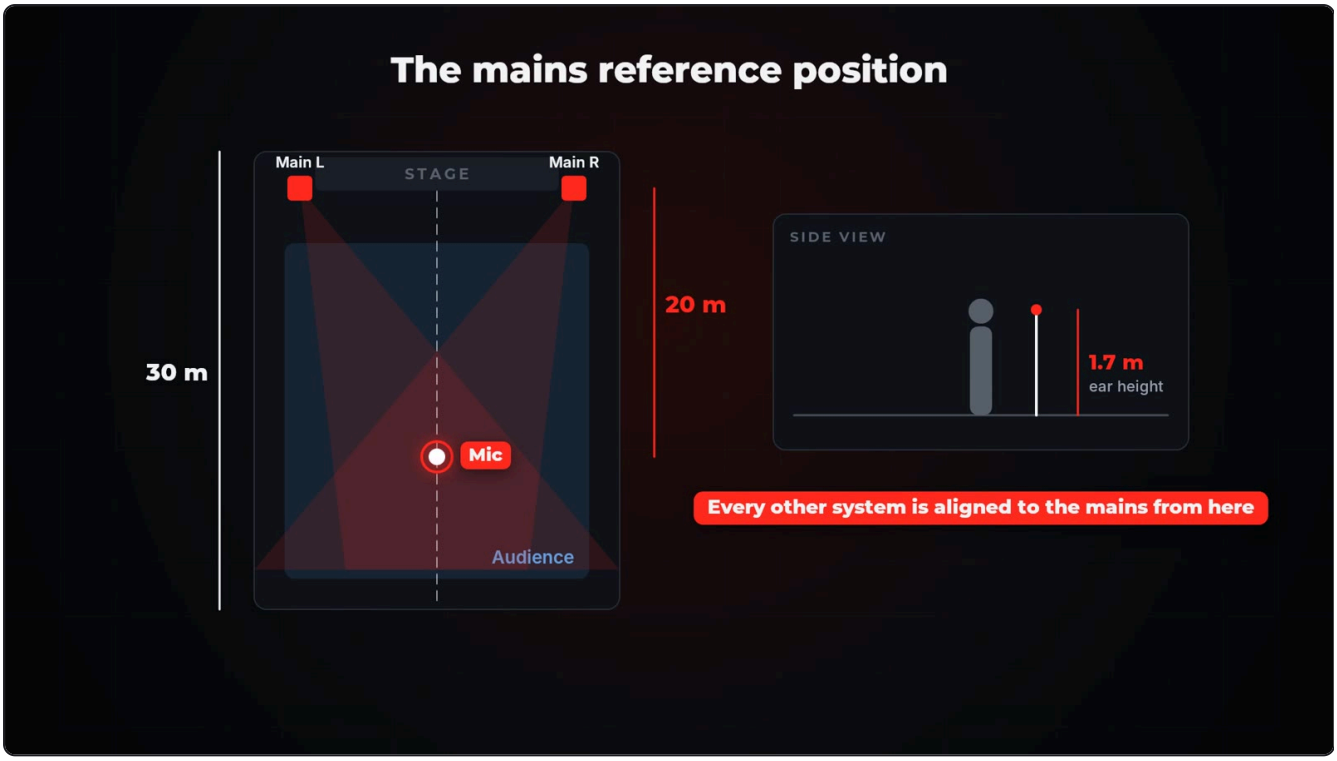


Figure 24. The mains reference position: every other system is aligned to the mains from here

How to turn a position into a measurement

1. Select a row in the table.
2. Press **Add as measurement**. A new, paused measurement named after the position is added. It uses the active measurement's input and reference.
3. Put the mic in place, tick the measurement in **Measurements** and press **D**.

Press **Save plan** to save the whole plan as a text file you can print or share with your crew.

General advice for every position

- Use one omnidirectional measurement microphone, the same one at every position. Point it at the source, or straight up if it is a random-incidence type (follow its data sheet).
- Put the mic at ear height and at least 1 m away from walls and other reflecting surfaces. Keep people out of the way during the measurement.
- Measure each system on its own first, then together. Mute the others at the processor or amplifier, not by pulling the signal. Always find the delay again after moving the mic.
- Check coherence before capturing. Above about 0.9 across the range you care about means a good position. If it is poor, move the mic half a metre to a metre and try again.
- Mark the position with tape or a note so you can return to the same spot after changes.

ALIGNING A CROSSOVER

The **Alignment > Crossover alignment** page finds the crossover region and the delay, polarity and all-pass settings that make two sources sum best.

Control	What it does	Range/default
Source A, **Source B**	The two sources: live measurements (*Live: …*) or captured traces with phase.	—
Find the crossover region automatically	Finds the region where both sources are equally loud.	On
Range	Manual region (when automatic is off). You can also drag the shaded region on the plot.	10–20000 Hz, default 60–160 Hz
Search	Largest delay to search.	1–500 ms, default 20 ms
Allow polarity inversion	Includes inverted polarity in the search.	On
Suggest all-pass filters (below 500 Hz)	Adds an all-pass option when the filters don't match.	On
Analyse	Runs the analysis.	—
Apply to trace	Adds the selected delay and polarity to the source trace.	—
Demo outputs A / B, **Apply to demo system**, **Demo processor…**	With the demo system: applies the suggestion to the virtual processor.	—

How to align subs to mains

1. Put the mic at the mains reference position (see *Planning microphone positions*).
2. Mute everything except the mains. Press **D**, then capture (**Space**) and rename the trace, for example *Mains*.
3. Mute the mains and play only the subs. You can limit the generator to about 30–200 Hz in **Generator options**. Press **D** and capture *Subs*.
4. Open **Crossover alignment**. Choose **Mains** as **Source A** and **Subs** as **Source B**.
5. Press **Analyse**.
6. Read the advice. For example: *"Delay Subs by 4.20 ms."* The text also shows how much the summation in the crossover region improves, and the phase difference at the frequency where both are equally loud.
7. Apply the selected option in your processor, then measure both together to confirm.



Figure 25. Crossover alignment: the suggestion, the predicted sum and the summation against delay

▶ Animated in the online manual

Reading the result

- The **Option / Polarity / Summation** table lists the best delay first, then other good choices and, if useful, an all-pass option. Select a row to plot it.
- The **Magnitude** plot shows A, B, **Sum now** (dashed), **Sum with suggestion** (white) and **Sum with all-pass** (green dashed).
- The **Phase** plot shows both phases in the crossover region, before and after the change.
- The **Summation** plot shows how well the sources sum for every delay applied to B (negative values delay A). The solid line is normal polarity and the dashed line is inverted. The green marker shows the selected option.



Figure 26. Solid: normal polarity. Dashed: inverted. Green: the suggestion

> **Tip:** When several delays sum almost as well, choose the shortest one that also keeps the arrival times close. Check the impulse to confirm.

If the advice says the phase slopes still differ by more than about 45°, the filters don't match. An all-pass filter or a crossover change may help.

Aligning fills and seams

For front fills, center fills, under-balcony fills and out-fill seams there is no crossover to find, because the fill and the mains overlap across the whole speech range. Turn off **Find the crossover region automatically** and set **Range** to about 250 Hz to 4 kHz. Then analyse the traces captured at the blend or seam position. Many engineers then add 5 to 10 ms so the image stays on the stage.

Applying the result

- **Apply to trace** adds the selected delay (and polarity) to the captured trace, so you can **Sum** the traces and see the result. It doesn't work for live sources or all-pass options: apply those in your processor and measure again.
- With the demo system, **Apply to demo system** writes the delay, polarity and all-pass settings into the virtual processor. Pi System Align guesses the matching demo outputs from the source names.

PLANNING ARRIVAL TIMES

The **Alignment > Arrival times** page lines up several sources from their measured arrivals.

1. For each source, play it alone at the listening position and press **Find delay** with the same reference.
2. Press **Add active measurement** to add the delay found for the active measurement, or **Add row** to type one in.
3. Fill in **Current delay (ms)** (the delay already set in your processor) and **Offset (ms)** (extra delay so fills and delays arrive a little after the mains).
4. Choose the source everything should line up with under **Align to**.
5. Press **Calculate**. **New delay (ms)** and **Change (ms)** are filled in, and a summary line lists the delays to set.

Column	Meaning
Source	Source name (editable).
Arrival (ms)	Measured arrival time.
Current delay (ms)	Delay set in the processor now.
Offset (ms)	Extra delay to add.
New delay (ms)	Delay to set.
Change (ms)	Difference from the current delay.

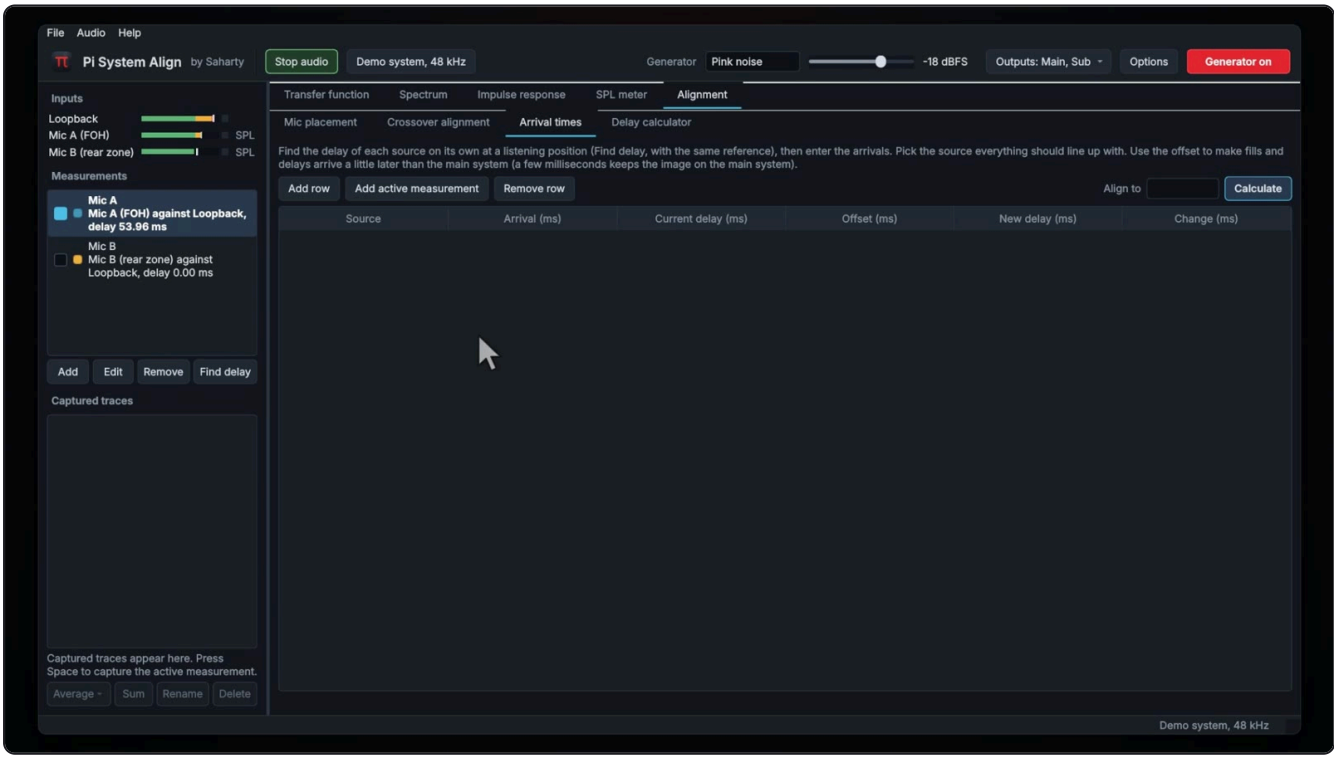


Figure 27. Arrival times: enter the measured arrivals and calculate the new delays

> **Tip:** A few milliseconds of offset on fills and delays keeps the image on the main system. For delay towers, 5 to 15 ms is a common range.

USING THE DELAY CALCULATOR

The **Alignment > Delay calculator** page turns distances into delay times. Set the **Air temperature** first (–20 to +50 °C, default 20 °C). The speed of sound is shown next to it and used everywhere, including the placement plan.

Section	Inputs	Result
Distance and time	**Distance** (0–2000 m, default 10 m); **Time** (0–5000 ms, default 10 ms)	Time for the distance; distance in metres and feet for the time
Path difference	**Listener to source A** (default 25 m), **Listener to source B** (default 18 m)	Which source arrives earlier, and the delay to set on it
End-fire sub array	**Boxes** (2–8, default 4), **Spacing** (0.1–10 m, default 1.0 m)	Delay for each box (box 1 is the rearmost)
Gradient (cardioid) sub pair	**Front to rear box** (0.1–5 m, default 0.6 m)	Delay for the rear box, which is also polarity inverted

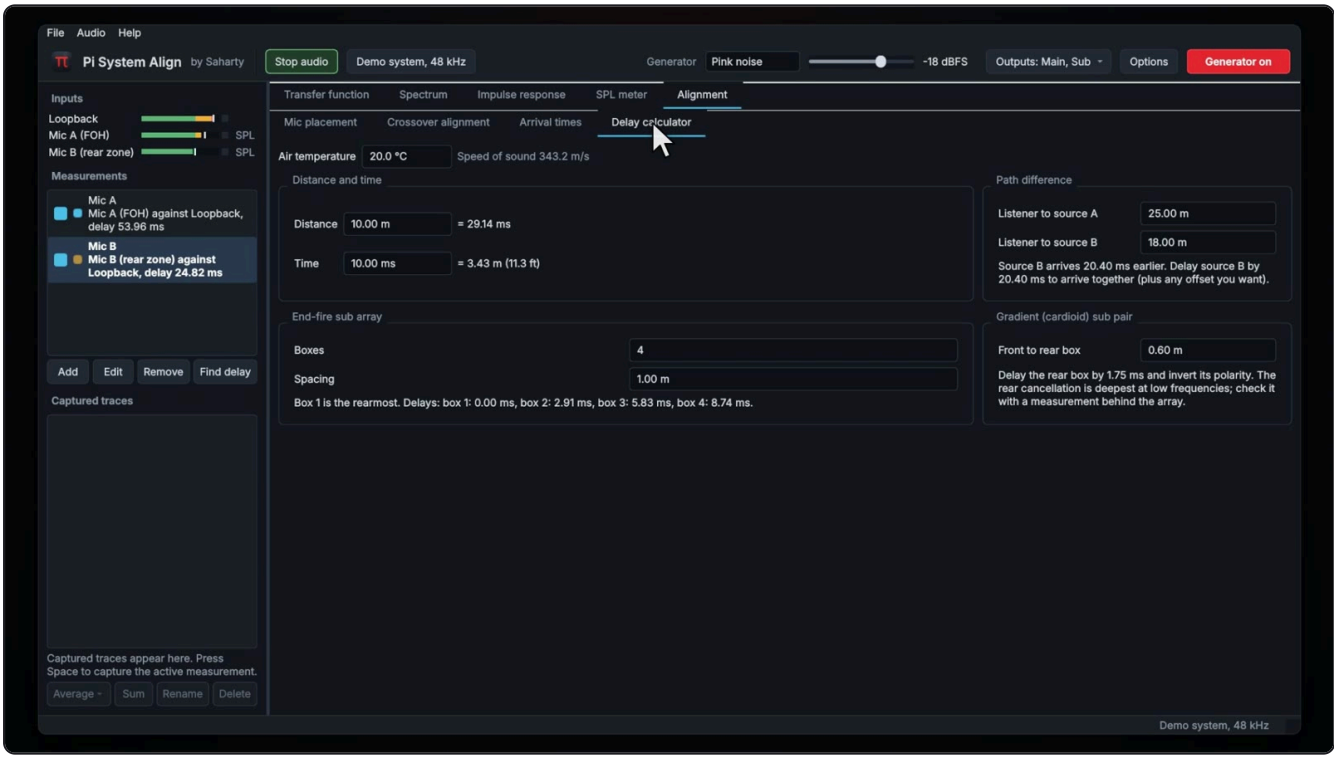


Figure 28. The delay calculator: distance and time, and path difference

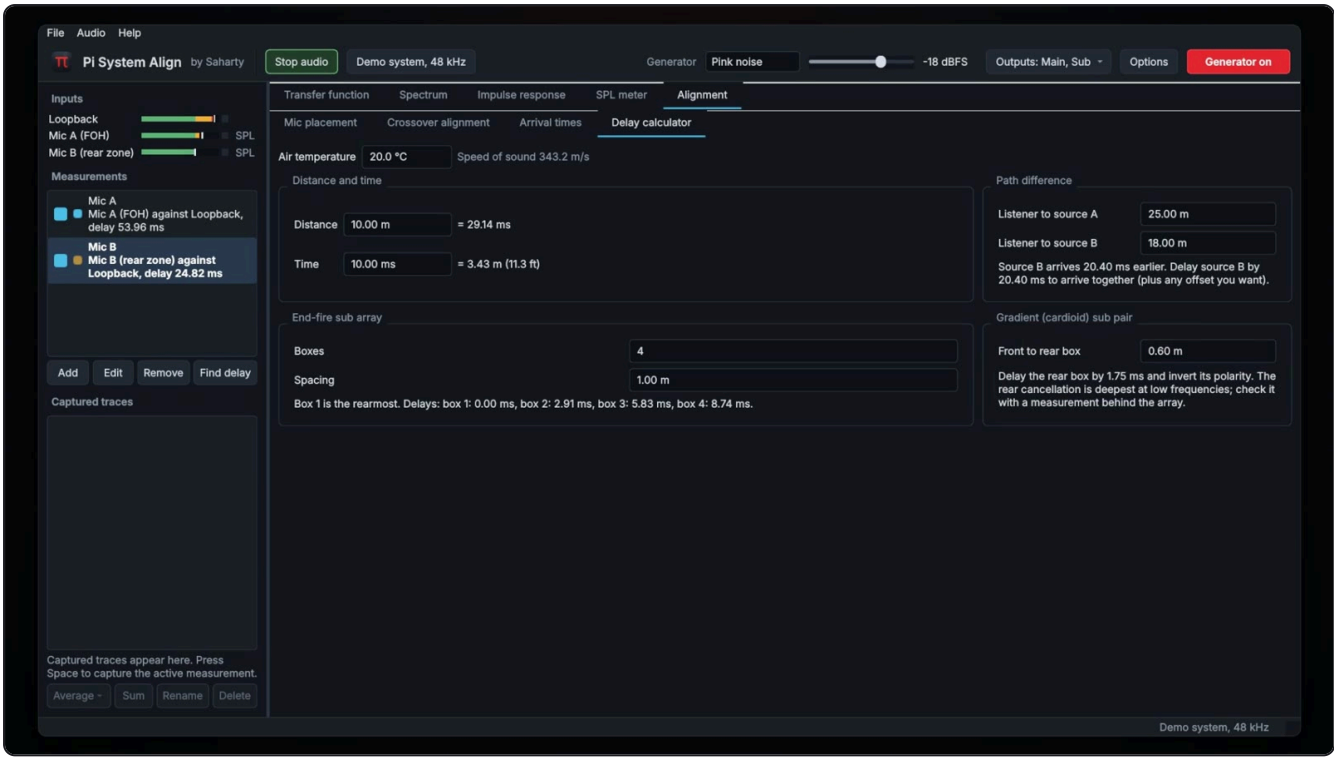


Figure 29. End-fire sub array delays calculated from the number of boxes and their spacing ▶ **Animated in the online manual**

> **Note:** A gradient pair cancels most at low frequencies behind the array. Check the result with a measurement behind the subs.

SESSIONS

A session (`.pisa` file) stores your settings, channels, measurements and all captured traces.

- **File > Save session...** (**Ctrl/Cmd+S**) saves the current session. Saving needs an active licence or the trial.
- **File > Open session...** (**Ctrl/Cmd+O**) opens a session. Opening always works, even in limited mode.

> **Tip:** Save a session after each venue. Next time, open it and compare today's traces with last time's.

SETTINGS & PREFERENCES

Pi System Align saves its state automatically when you quit, and restores it at the next launch. This includes:

- the audio device, sample rate, block size and channel counts;
- the input names, calibration and microphone corrections, stored separately for each device;
- the measurements and the generator outputs for each device;
- the transfer function, spectrum, impulse response and SPL meter settings;
- the air temperature and the mic placement plan;
- the window size, sidebar width and the open tab;
- the demo system processor settings.

Because inputs and measurements are stored for each device, switching between the demo system and your interface brings back each device's own setup.

Where	Setting
Audio settings	Device, sample rate, block size, channels
Generator options	Band limits, sine frequency, sweep length
Input settings	Name, calibration, mic correction
Each tab's toolbar	Display and analysis settings
Delay calculator	Air temperature (also used by the placement plan)

KEYBOARD SHORTCUTS / REMOTE CONTROL

Key	Action
Space	Capture the active measurement (on the **Spectrum** tab: capture the spectrum)
D	Find the delay of the active measurement
G	Generator on or off
R	Reset averages
Ctrl+1 to **Ctrl+5**	Switch views: Transfer function, Spectrum, Impulse response, SPL meter, Alignment
Ctrl+O	Open a session
Ctrl+S	Save a session

On a Mac, use **Cmd** in place of **Ctrl**. You can also see the list from **Help > Keyboard shortcuts**.

Pi System Align has no OSC, MIDI or timecode remote control.

TIPS FOR LIVE/STUDIO USE

- **Follow a simple workflow for every show:** connect a loopback and your mic and check the levels; calibrate the mic; plan the mic positions; measure the mains and tune them first; align the subs at the crossover position; align fills and delays with arrival times; then walk the room, measure and listen.



Figure 30. A simple workflow for every show

- **Use a loopback whenever you can.** It includes the latency of your processor and converters, so arrivals and delays are measured from the real signal path.
- **Find the delay again every time you move the mic.** The phase and coherence only read correctly with the right delay compensation.
- **Watch coherence before you trust a trace.** Turn on **Blank coherence below** (for example 50%) to hide unreliable data, especially when you measure with programme material.
- **Use more averages for decisions.** 16 or more averages give a steady trace for capturing. Fewer averages react faster while you adjust things.
- **Measure with show music.** The transfer function works with any broadband signal, so during sound check or the show you can keep measuring with programme as the source, without playing noise.
- **Limit the noise for subs.** In **Generator options**, set **Limit the noise to a band** to about 30–200 Hz when you measure subs. The energy goes where the subs work, and the rest of the system is spared.
- **Check the live impulse for polarity.** A negative main peak means the polarity is inverted somewhere. Fix that before you judge a crossover.
- **Track delays outdoors.** Temperature and wind change arrival times. Turn on **Track** for the measurement and set the **Air temperature** in the delay calculator.
- **Keep an eye on Dropouts.** If the status bar shows dropouts, choose a larger **Block size** in **Audio settings**.

- **Practise first.** The demo system processor responds just like a real one. Try every tool there before the doors open.

TROUBLESHOOTING

Symptom	Cause	Fix
No audio devices are listed	The interface is not connected or is in use by another program	Press **Rescan devices**. Close other audio programs. On Windows, try a different driver type.
The interface doesn't appear on Windows	No ASIO driver is installed	Install the manufacturer's ASIO driver, or use the interface's WASAPI entry.
Mac: inputs show no signal	Microphone access was not granted	**System Settings > Privacy & Security > Microphone**: allow Pi System Align.
"No clear arrival found"	The generator is off, no outputs are selected, the mic has no signal or the reference doesn't carry the generator	Turn on the generator (**G**), choose its **Outputs**, check the mic input meter and check that the reference carries the signal.
"The generator has no outputs selected"	No outputs are ticked under **Outputs**	Open **Outputs** and tick the outputs that feed the system.
Outputs shows *"Start audio to choose outputs"*	Audio is stopped	Press **Start audio** first.
Phase is a mess and coherence is low	The delay was not found, or the mic is in a null or reflection	Press **D**. Move the mic half a metre to a metre and try again.
The trace reads far too low before **Find delay**	The delay compensation is not set yet	Press **D**. The scale fits itself once the delay is found.
Peak shown as negative after **Find delay**	Polarity is inverted somewhere in the path	Check the cabling and processor polarity before aligning.
Calibration warns that the tone is not at 1 kHz	The calibrator is off or not seated properly	Re-seat the calibrator and measure again.
Calibration warns of clipping	The preamp gain is too high	Lower the gain and calibrate again.
SPL meter shows dBFS	The input is not calibrated	Calibrate it with **Calibrate...**.
T20 or T30 shown in amber	The decay is not a straight line (noise or a short sweep)	Use a longer sweep, more averages or a quieter room.
Long sweep lengths are greyed out	The sweep doesn't fit the audio buffer at this sample rate	Choose a shorter sweep.
Crossover alignment: *"Both sources need data"*	A live source has no data or the traces are missing	Start audio for live sources, or capture traces of both.
Apply to trace: *"That source is a live measurement"*	Changes can only be applied to captured traces	Apply the change in your processor and measure again.
Sum or **Average** refuses	Spectrum and transfer traces are mixed, or a trace has no phase (for **Sum**)	Select traces of the same kind, with phase for a sum.

Symptom	Cause	Fix
Dropouts appear in the status bar	The audio buffer is too small for the computer	Choose a larger **Block size**.
Buttons read **Activate to save**	Limited mode: no active licence or trial	Open **Help > Licence...** and activate, or start the trial.
Banner: *the licence system is unavailable*	The licence component is missing	Reinstall Pi System Align.

SPECIFICATIONS & SYSTEM REQUIREMENTS

Specifications

Item	Specification
Current version	1.1.0
Measurements	Transfer function (MTW or fixed FFT 4096–65536), spectrum, impulse response, SPL
Alignment	Mic placement, crossover alignment, arrival times, delay calculator
Systems in the placement plan	Mains, subs, center fill, front fills, out fills, under balcony, delays
Traces	Capture, power/complex/dB average, complex sum, rename, edit, CSV import/export
Generator	Pink noise, pink periodic, white noise, sine, log sweep
Sample rates	44.1, 48, 88.2, 96 kHz
Channels	Up to 64 inputs and 64 outputs
Spectrum banding	1/1 to 1/48 octave, A/C/Z weighting
Room acoustics	EDT, T20, T30, C50, C80, D50, Ts; broadband and 63 Hz–8 kHz octaves
SPL	A/C/Z, Fast/Slow/Leq, max, peak, Leq 1/10/15 min, one-hour per-second history
File formats	Sessions `.pisa`; traces `.csv` / `.txt` / `.frd`; impulse responses `.wav`; licence `.pilic`
Audio drivers	Core Audio (macOS), ASIO and WASAPI (Windows)

System requirements

macOS

	Minimum	Recommended
OS	macOS 11 Big Sur	Latest macOS
Processor	Intel or Apple silicon	Apple silicon or recent Intel
Memory	4 GB	8 GB
Disk	300 MB	300 MB
Audio	2-channel interface + measurement mic	Multichannel interface + calibrated omni mic

Windows

	Minimum	Recommended
OS	Windows 10 64-bit	Windows 11 64-bit
Processor	Intel / AMD, 64-bit, 2 cores	Intel Core i5 / AMD Ryzen 5
Memory	4 GB	8 GB
Disk	300 MB	300 MB
Audio	2-channel ASIO interface + measurement mic	Multichannel ASIO interface + calibrated omni mic

Pi System Align works with Core Audio interfaces on macOS, ASIO interfaces on Windows, USB digital consoles, and MADI or network audio cards through their drivers. The built-in demo system needs no interface. Linux is not supported.

SUPPORT

- **Email:** support@pi-ware.com
- **Website:** www.pi-ware.com (downloads, your licences and the account page)
- **Tutorials:** youtube.com/@pibysaharty

When you contact support, include your version number (**Help > About Pi System Align by Saharty**), your operating system and your audio interface.

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www.pi-ware.com · support@pi-ware.com

Video tutorials: youtube.com/@pibysaharty

Online manual with animations: https://www.pi-ware.com/manuals/pi-system-align/

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