SMartin

Audio Measurement Suite by Cana San Martin

V. 5_2_154

User Manual

SMartin - Audio Measurement Suite (v5.2)

1. Introduction, scope and fundamentals of measurement

1.1 Purpose of the software

SMartin is an electroacoustic measurement suite designed for **real-time analysis**, **system tuning**, **room verification**, **level control**, and **program monitoring**. The software's objective is to convert audio signals (measurement and reference) into interpretable **quantitative** and **graphical indicators** for informed technical decisions: tonal corrections, time alignment, room behavior assessment, target verification, and loudness/peak control.

In methodological terms, **SMartin** works in two complementary domains:

Frequency domain (e.g., spectrum/RTA, frequency response).

Time domain (e.g., level history, averages, measurements by integration/window).

The tool is organized as a set of modules (tabs) that cover these typical domains and workflows. The interface includes tabs such as **RTA + Spectrogram**, **Transfer function**, **Generator**, and verification and room modules (Room parameters, Acoustic graphs, Level history, Wizards), as well as **Studio monitor (Loopback)** for desktop audio monitoring.

The free use of this software is limited to students in audio-related fields. For professional use, you must request permission from the developer. You can also make a symbolic donation to the FIAPLI Foundation, which aims to develop inclusive technologies for people with disabilities.

1.2 Organization by modules

SMartin adopts a modular architecture (tabs), where each module implements a specific task within a measurement flow. The visible structure includes, among others, the following main modules: **RTA + Spectrogram** , **Transfer function** , **Generator** , **Studio monitor (Loopback)** , **Room parameters** , **Acoustic graphs** , **Level history** , and **Wizards** .

At the reporting level (export/diagnostics), the system includes outputs organized into blocks such as **QA/Summary** , **Wizards Summary** , **Room Acoustic Parameters** , **Acoustic Graphs** , and **Level History (SPL)** , consistent with the "measurement → interpretation → verification" approach. Each graph can be exported as an image or data file, and there is a global image and text capture tool that prints diagnoses and screenshots in PDF format.

Practical interpretation of each module (overview):

- **RTA + Spectrogram** : Spectral analysis to identify tonal balance, resonances, background noise, band accumulation, and averaging/holding behavior. Useful tools for complementary interpretation of the graphs (peak frequency, deltas, target curves, etc.).

- **Transfer function** : comparative reference/measurement analysis to estimate system behavior (frequency response and phase/coherence correlates according to implementation), geared towards alignment and equalization based on criteria. Ability to calculate and export impulse response.

- **Generator** : stimulus signal (e.g., noise/tones depending on system configuration) to excite the system under test in a controlled manner.

- **Studio monitor (Loopback)** : Monitoring of audio played by the computer (desktop audio) with typical studio tools (RTA, goniometry, correlation, level metrics/LUFS). It is supported on most current operating systems.

- **Room parameters / Acoustic graphs** : room parameters and useful representations for acoustic evaluation, derived from the transfer function.

- **Level history** : temporal/statistical monitoring of levels (SPL/dBFS depending on calibration and mode). Export capability for expert analysis.

- **Wizards** : guided assistants for recurring tasks (e.g., checklists and structured procedures, adjusting sub bass, fills, etc.).

1.3 Essential concepts and units of measurement

To operate the software rigorously, it is essential to differentiate **what magnitude is being measured** , **in what unit** , and **with what reference** .

1.3.1 dBFS vs dB SPL

- **dBFS (decibels Full Scale)** : a digital unit relative to the maximum representable in the system (0 dBFS is the ceiling). Anything exceeding 0 dBFS is not representable (clipping).

- **dB SPL (Sound Pressure Level)** : acoustic sound pressure level (referred to 20 μPa). To obtain SPL from a digital chain, **calibration is required** : microphone + preamplifier + conversion + offset/gain adjustment.

Operational implication: Without calibration, SMartin delivers relatively consistent measurements (dBFS), but cannot guarantee direct equivalence with dB SPL.

1.3.2 Measurement signal vs reference signal

In system measurements, it is common to work with **two signals** :

- **Reference (Ref)** : what “enters” the system (the signal that is sent).

- **Measurement (Meas)** : what “comes out” of the system (captured by microphone or measurement point).

This approach allows for robust comparative analyses (e.g., transfer function), provided that:

- both signals are correctly routed,

- sufficient temporal synchronicity exists,

- there is no significant crosstalk between channels,

- the system operates at a stable and consistent **sample rate .**

1.4 Fundamentals of real-time spectral analysis

The core of modules such as RTA and spectrogram relies on spectral transforms (typically FFT), where a time signal (x[n]) is converted into a set of frequency components.

1.4.1 Frequency resolution vs. time resolution

In practical terms:

- Larger **FFT/window size** means better frequency resolution (thinner bands), but worse time resolution (more read latency and slower averages).

- At a lower FFT, the reading reacts faster, but with less frequency detail.

This explains why a “fast” RTA can be useful for transient events (hits, consonants) while a “fine” RTA is more appropriate for steady-state tonal balance (pink noise, sustained music).

1.4.2 Ventaneado (windowing)

Windowing reduces discontinuity effects at the edges of the analyzed block and controls the compromise between:

- **leakage** (spectral leaks),

- **main lobe width** (resolution),

- **side lobe behavior** (interband contamination).

In terms of operation: choosing the window and FFT is not a mere aesthetic detail; it modifies the spectrum reading and stability. By default, the system works with a stable and accurate Hann window.

1.4.3 Integration, averages and “holds”

Modern RTAs combine:

- **Temporal integration** (how much the reading "averages").
- **Peak hold** .
- **Average hold** (energy holding/average).
- **Smoothing** (smoothing by band or fraction of an octave).

The key is to interpret the correct indicator for each task:

- Peak hold helps to "see" resonances that appear with specific events.
- Average hold helps to estimate typical program energy.

1.5 Preparing a measurement: technical recommendations

The following are practical criteria for obtaining reproducible results.

1.5.1 Measurement microphone: selection and placement

Recommended type: omnidirectional measuring microphone, with known response and preferably calibratable.

Location (general criteria):

- Avoid extreme proximity to reflective surfaces (walls, tables) that induce comb filtering.
- For "room balance", choose representative positions (not just the most favorable point).

- For alignment or adjustments by channel (sub/top), place the microphone in an area where both contributions are significant and with a good signal-to-noise ratio.

Guidance: Follow the microphone calibration criteria (0°/90°) according to the correction file/curve if applicable.

1.5.2 Speaker/system under test and stimulus

For controlled measurement, use a stable stimulus (e.g., noise/tone from the Generator). The goal is:

- increase repeatability,
- ensure sufficient SNR,
- avoid bias in musical content (which is highly variable).

Good practices:

- Work at levels sufficient to overcome ambient noise, but without entering into unwanted compression/limitation.
- Prevent dynamic processing (limiters, compressors) from altering the reading while adjusting tonal response; if this is not possible, at least keep it under control and document it.

1.5.3 Profit and "headroom" (digital chain)

To maintain integrity:

- Avoid clipping at any point.
- Maintain peaks with headroom for transients.
- Adjust the microphone preamp gain so that the system doesn't "breathe" in the background noise but also doesn't saturate.

1.6 General interface: menus and navigation

The application is organized with typical top menus: **File** , **Audio I/O** , **View** , **Help** , and **About** , consistent with the need to operate audio settings, display, and help/documentation.

The logical flow of use is usually:

- Configure audio in **Audio I/O** (devices, channels, API).
- Confirm that there is a valid signal (reasonable levels).
- Open the module of interest (RTA, Transfer Function, Studio monitor, etc.).
- Adjust analysis parameters (resolution, integration, window, smoothing, etc.).
- Interpret with criteria (compare positions, averages, consistency/stability).

1.6.1 Controls

- **Left click:** parameter selection.
- **Left click and hold** : zoom, scroll in graphics.
- **Multiple left clicks:** stress and anxiety release.
- **Right-click:** drop-down menu for exporting graphics.

1.6.2 Requirements

- **Operating System:** Windows 11 (the Loop Back function may not work correctly on earlier operating systems).
- **Minimum recommended RAM** : 8 GB.
- **Microprocessor:** Intel i5 and above.
- **Disk storage:** 150 MB + space available for exports.
- **Audio Engine:** WDM. While it supports multiple drivers, including ASIO engines, the most stable way is to route the ASIO or other driver to the operating system's input and output and select

1.7 Recommended workflow: Scientific "Quick Start"

This procedure summarizes a typical startup process geared towards reliable results:

1. **Define the measurement objective.**
   Examples: system tonal balance, sub-bass verification, alignment, room analysis, studio loudness monitoring.

2. **Stabilize the system.**
   Set the sample rate, disable automatic changes, and verify that the driver is not forcing conversions.

3. **Correct channel routing**
   . Identify unambiguously:

   a. Measurement channel (microphone or capture point).

   b. Reference channel (if applicable).

4. **Adjust gain and verify integrity**
   . No clipping, with sufficient SNR.

5. **Select the appropriate module**

   a. RTA for spectrum.

   b. Transfer function for comparison ref/meas.

   c. Studio monitor for desktop audio and program metrics.

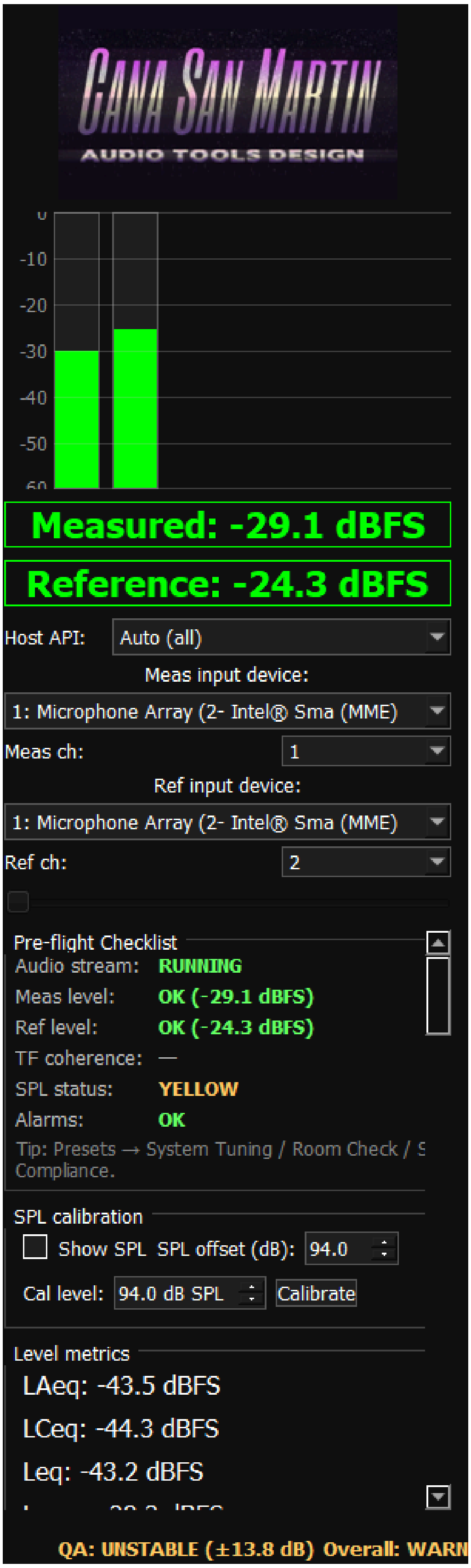
6. **Configure parameters based on the phenomenon**

   a. Adjust integration/FFT/window depending on whether you are looking for transients or steady state.

   b. Use smoothing judiciously (don't "hide" problems with excessive smoothing).

7. **Repeat and validate.**
   A single measurement is rarely conclusive: repeat in relevant positions and compare trends (averages or multi-position) before making corrections.

2. Right column - Control panel, reference and status



The **right-hand column** of **SMartin concentrates all the instant, contextual, and status** information of the measurement. Its function is not to "analyze" (that happens in the main graphs), but **to inform, validate, and give operational confidence** to the user while they are measuring.

It can be read from **top to bottom** as a logical chain:

> *Is a signal coming in? → From where? → At what level? → Is the measurement valid? → Am I within limits? → Is it being recorded?*

2.1 . Input Level Meters (Vertical VU)

Two vertical bars:

- **Measured** : RMS level of the measurement signal (microphone/main input).
- **Reference** : RMS level of the reference signal (loopback, console, ref in).

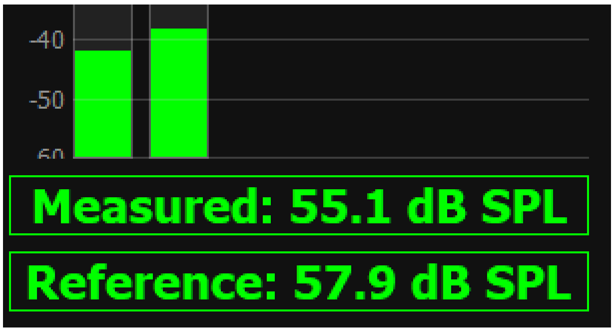
The values are expressed in **dBFS** (decibels relative to full scale digital).

1. They represent the **average energy level** of each signal.
2. They allow you to verify:
   a. Signal presence.
   b. Relative balance between measurement and reference.
   c. Headroom available before clipping.
3. Typical working values:
   −60 to −30 dBFS (depending on microphone, preamp and environment).
4. Large differences between *Measured* and *Reference* usually indicate:
   a. Incorrect routing.
   b. Unbalanced profits.
   c. Contaminated or saturated reference.

2.2 . Numerical Reading Measured / Reference

The numerical values are shown below the VU values:

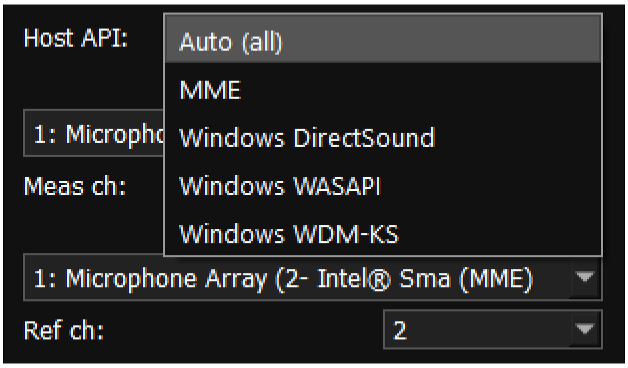
Measured: 55.1 dB SPL Reference: 57.9 dB SPL



Contribution to the VU:

- Absolute precision.
- Ability to repeat setups (compare sessions).
- Useful for calibrations and documentation.

2. 3. API Host Selector



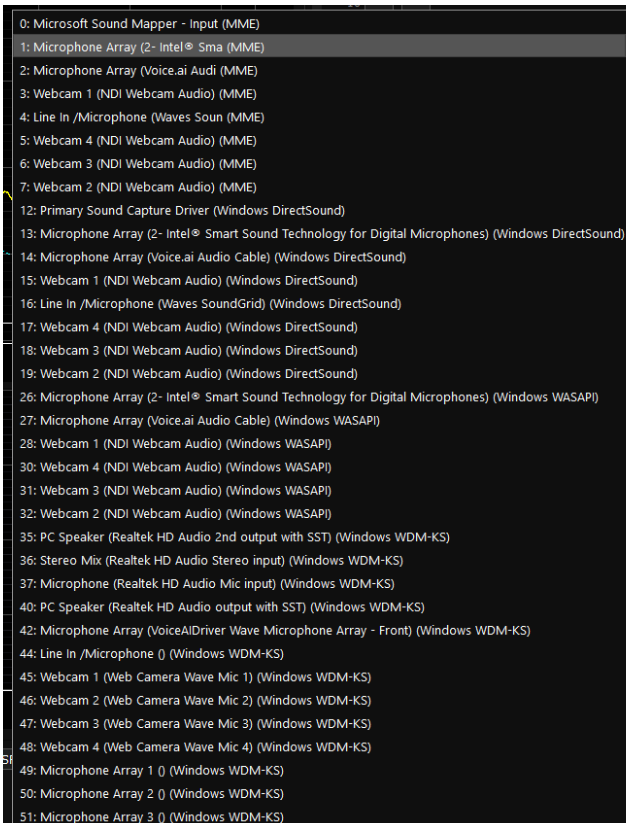
Dropdown menu with **active audio API** :

- Auto (all)
- AT THE HOUSE
- WDM-KS
- ASIO (if the backend supports it)
- MRS., etc.

Determine **which PortAudio layer** is used to enumerate and open devices.

- It affects **latency** , **channel availability** , and **stability** .
- It is critical in multichannel systems or transfer measurements.

2.4 . Device and channel selection



Modules

- **Meas input device**
- **Meas ch**
- **Ref input device**
- **Ref ch**

They allow you to explicitly select:

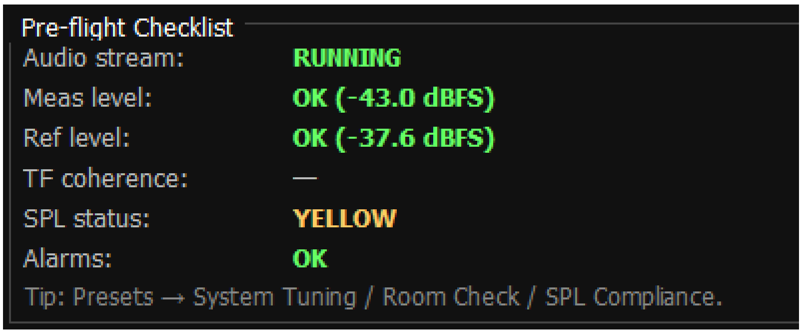
- **Physical device** (interface, integrated microphone, virtual cable).
- **Specific channel** within that device.

SMartin always works with **discrete channels** , not forced pairs. This allows:

- Using a single channel from a multi-channel interface.
- Measure L and R separately.
- Configure internal references without duplicating hardware.

2.5 . Pre-flight Checklist (Measurement Validation)

This block summarizes the **logical state of the measurement** :



- **Audio stream**: RUNNING / STOPPED
- **Meas level**: OK / LOW / CLIP
- **Ref level**: OK / LOW / CLIP
- **TF coherence** (if applicable)
- **SPL status**
- **Alarms**

It is a **rapid diagnostic expert system** that allows you to detect common errors **before** relying on the graphs, such as:

- Microphone disconnected.
- Incorrectly routed reference.
- Unusable levels.
- Conditions not valid for TF or SPL.

2.6 . SPL (Traffic Light) status indicator

A traffic light-type visual indicator:

- 🟢 Green: within parameters.
- 🟡 Yellow: Caution.
- 🔴 Red: outside configured limits.

Depending on the configuration, it can be used to evaluate:

- LAeq
- Lmax
- Regulatory thresholds
- Active alarms

Typical use

- Regulatory compliance control.
- Real-time monitoring during events.
- Evidence for reports.

2.7 . SPL Calibration

Controls:

- **Show SPL**

- **SPL offset (dB)**
- **Cal level**
- **Calibrate**

The software measures digital amplitude in **dBFS** (decibels relative to full scale). To estimate **dB SPL** , a very simple conversion based on an offset is used:

That **offset** absorbs everything: microphone sensitivity, preamp/interface gain, etc.

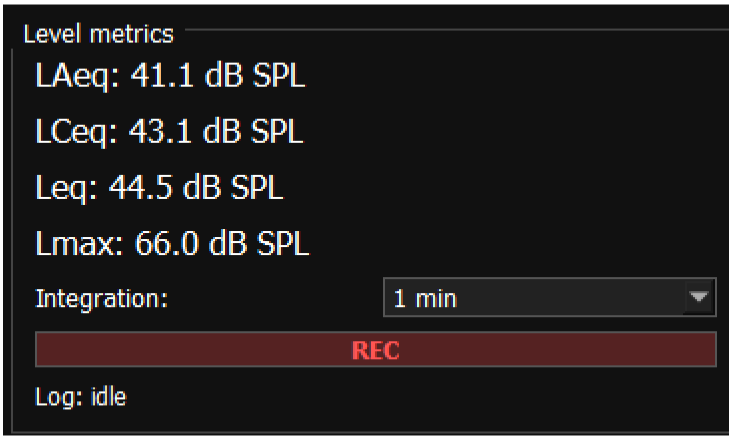
Typical procedure

1. **Choose the correct input** (Measurement input device and its channel). If the routing is incorrect, the calibration will not work because it will be calibrating a different channel.
2. **Connect an acoustic calibrator** to the microphone (ideally: 94 dB SPL @ 1 kHz or 114 dB SPL @ 1 kHz).
    a. Make sure there is no AGC/compressor in the interface or in Windows.
    b. Adjust the preamp to avoid clipping (the important thing is a clean signal, not a "high" one).
3. In **SMartin** , set *the Cal level* to the calibrator value (typical: **94 dB or 114 dB**).
4. Wait for the level to stabilize (stable integration mode helps).
5. Tap **Calibrate** .
    a. **SMartin** takes the measured level (dBFS) and sets the **Offset** to match the Cal level .
6. Enable Show SPL to have the software display readings in dB SPL (where applicable).

Good practices

- If you change the preamp gain, your offset changes. Recalibrate.
- SPL calibration is the "anchor point" (typically 1 kHz). For spectral accuracy, FR calibration should also be added.

2.8 . Level Metrics (Energy Statistics)



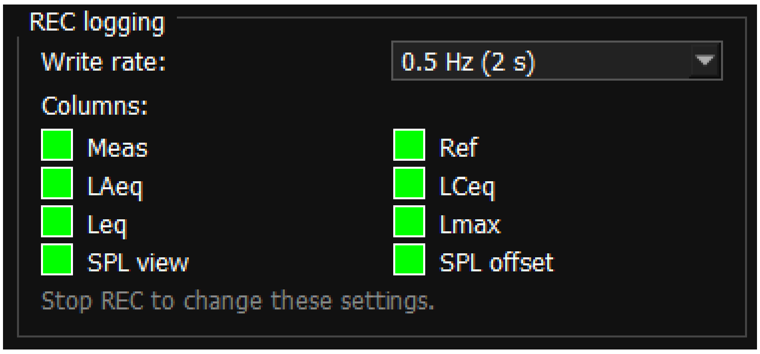
Metrics shown

- **LAeq** – A-weighted energy (human perception).
- **LCeq** – C-weighted energy (broadband).
- **Leq (Flat)** – Unweighted energy.
- **Lmax** – Maximum recorded peak.

They are **time integrals** , not instantaneous values.

- They represent cumulative sound exposure.
- They are the basis for regulations, reports and impact analyses.

2.9 . Temporary integration and logging



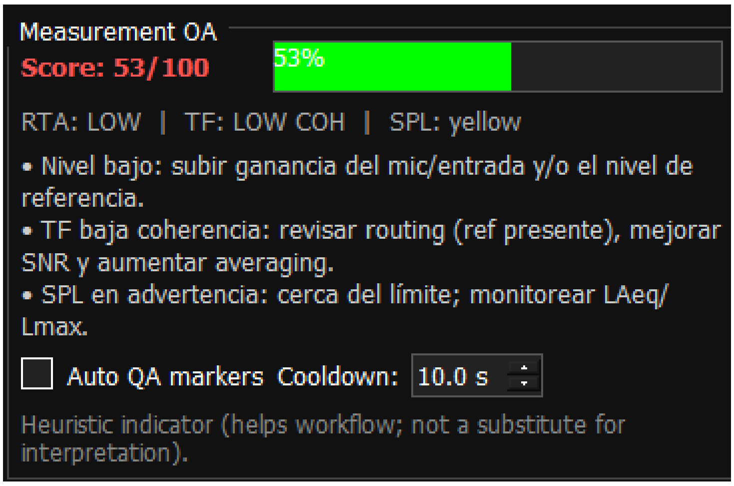
Controls

- **Integration** (not 1 min)
- **REC**
- **Log status**
- **Write rate**
- **Column selection**

What they do:

- They define **how often** the values are calculated.
- They allow **data to be recorded in real time** .
- They generate traceable files for reports.

2.10 . Scroll area and extended modules



The bottom part of the column is **scrollable** and contains:

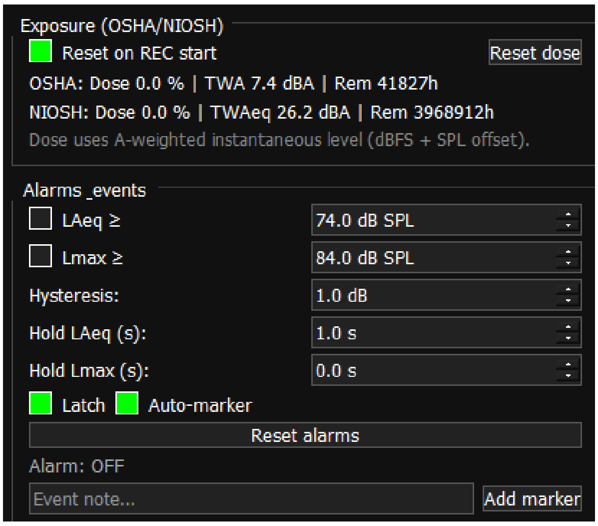
- Advanced alarms.
- Events and scoreboards.
- I/O Profiles.
- Regulatory configuration.
- Future extensions.

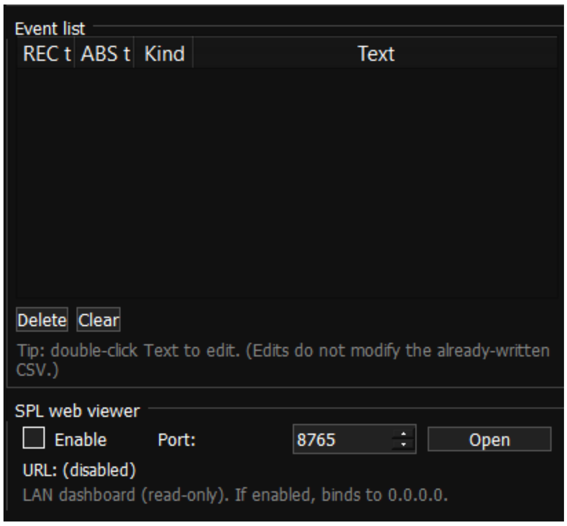
This ensures that **the column does not grow indefinitely** , always keeping the critical modules visible.

The right-hand column is not "decorative": It is the **semantic control center** of **SMartin** :

- The **graphs show** .
- The **right-hand column validates, contextualizes, and certifies** .

If the graphs are the “oscilloscope”, the right column is the **instrument panel of the measuring system** .





2.11. "RTA Pro Utilities" Window

RTA Pro Utilities is the "pro" module for operating the RTA as a tuning tool (systems engineering style): **targets** , **tolerances** , **comparison** , **traces** and **simultaneous multi-RTA** , plus **FR calibration** .

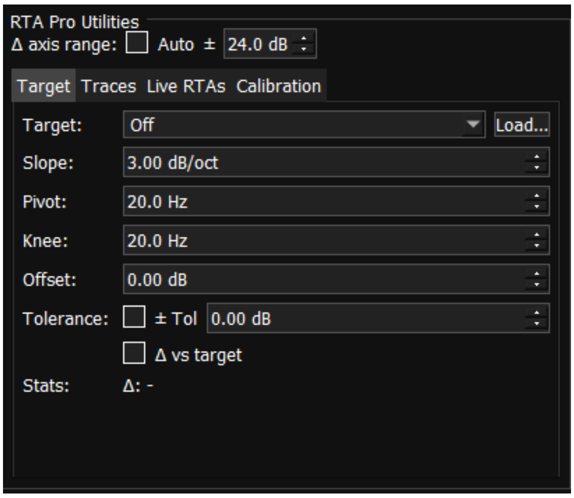
2.11.1 Control "Δ axis range" (Delta axis range)

Above the tab there is a control:

- `Δ axis range:`
- `Auto`
- `± [valor] dB`

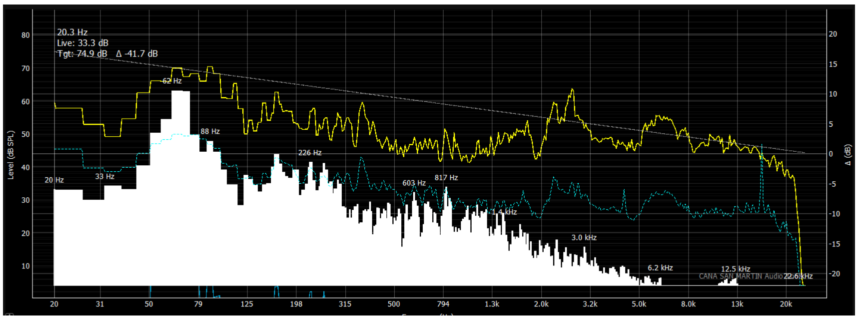
When you activate deltas (e.g., **Δ vs target** or **Compare Δ (Live – Trace)**), **SMartin** draws a "delta" curve that represents a difference in dB.

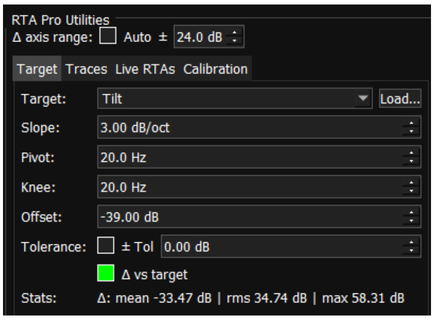
- The delta makes sense on a **secondary Y-axis** (right), so that it does not depend on the large range of the RTA.
- `Auto` : Autoscales the delta axis according to what is being viewed.
- `± X dB` : sets the range (for example, ±6 dB, ±12 dB, ±24 dB).
  This ensures consistent readings (very useful for comparing captures at different times).



2.11.2 Tab "Target"

Target **defines** a "target curve" to compare the RTA.





Main controls:

- `Target:` Off / Flat / Tilt / HF Tilt / Custom (+ botón `Load...`)
- `Slope` (dB/oct)
- `Pivot` (Hz)
- `Knee` (Hz)
- `Offset` (dB)
- `Tolerance:` checkbox `± Tol` + valor (dB)
- `Δ vs target` (muestra la delta Live – Target)
- `Stats:` (delta summary)

Types of target:

- **Off**: sin target.
- **Flat** : a flat reference. Useful for linearity measurements or for comparison with an average level.
- **Tilt** : a straight line with a slope on a logarithmic scale (dB per octave) around a **pivot**
  .
  It is typical for a conceptual "house curve" (trend).
- **HF Tilt** : "Tilt" only from a **Knee position** (below it remains flat).
  This is useful for lenses where you don't want to adjust bass/midrange frequencies but do want to focus on treble.
- **Custom** : target loaded from file (CSV/TXT of 2 columns: frequency and dB).

Parameters

- **Slope (dB/oct):** how many dB the target changes per octave.
  In SMartin, a positive slope corresponds to the typical behavior: **higher frequency, lower target level** (downward tilt in HF).
- **Pivot (Hz):** Frequency where the tilt is "anchored". It is the reference point for the tilt.
- **Knee (Hz):** HF tilt starts at this frequency (if the target is HF Tilt).
- **Offset (dB):** moves the target up/down.

Tolerance (± Tol)

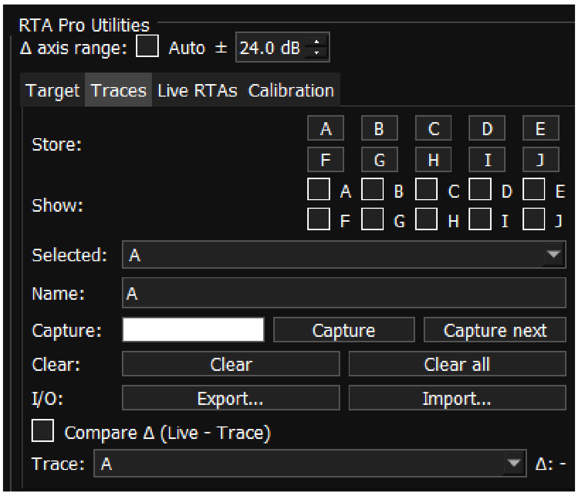
- `± Tol` draws a band around the target (e.g., ±3 dB).
This allows you to visually read "meets/does not meet" without having to interpret micro-variations.

Δ vs target

- When you activate `Δ vs target` , a delta curve (Live – Target) appears and the "Stats" panel is updated.
- Ideal for adjustment: you want the delta to be "centered" and within tolerance.

2.11.3 Tab "Traces"

This tab is used to **capture** the RTA spectrum and **compare it** .



In *"Store / Show"* there is a matrix of slots A–J :

- **Store:** A..J buttons (save the current spectrum)
- **Show:** checkboxes A..J (show/hide each trace on the RTA)

Typical use:

- "Before" captures in A, EQ adjustments, "After" captures in B.
- Sample A and B, and compare.

En *"Selected / Name / Color / Capture / Clear / I/O"*

- **Selected:** Choose which slot you are editing (A..J)
- **Name:** descriptive name (FOH, Balcony, Sub array, etc.)
- **Color…:** assigns color to that trace (so that the overlay is readable)
- **Capture / Capture next:** Capture to the selected slot or the next available slot.
- **Clear / Clear all:** clears one or all
- **Export… / Import…:** saves/loads traces to/from file (for sessions, reports, comparisons)

Compare Δ (Live – Trace)

- `Compare Δ (Live - Trace)` : activates delta against the chosen trace.
- `Trace:` selector of the trace to compare
- `Δ:` stats (summary of difference)

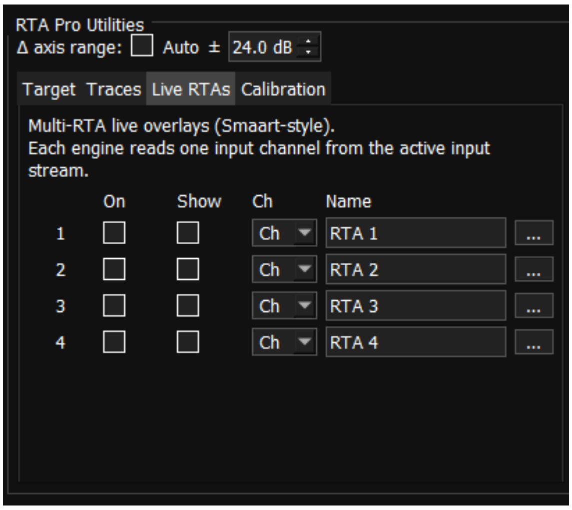
This makes RTA a QA tool:

- "How much did it change compared to the reference point?"
- "Am I within the margin?"

2.11.4 Tab “Live RTAs”

This part implements the concept of “simultaneous multi-RTA” (several live spectra superimposed), useful if you have multi-channel interfaces.

What it allows:



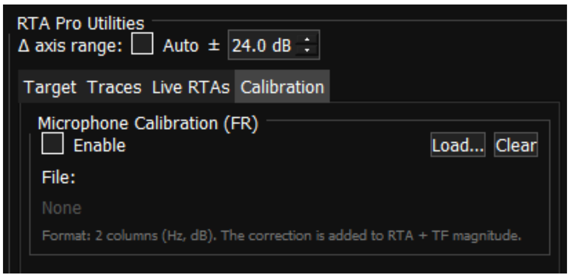
1. Configure up to **4** additional live spectra (each with):
    a. Enable
    b. Show
    c. Channel
    d. Name
    e. Color

Typical use:

- Quickly compare several microphones **(FOH vs side fill vs stage)** .
- View spectral differences between points without having to "save trace".

2.11.5 Tab “Calibration”

Microphones (and the signal chain) have frequency response errors. The FR calibration is a curve:



- Column 1: **Frequency (Hz)**
- Column 2: **Correction (dB)**

That **SMartin** interpolates and **adds** to:

- the **RTA** (on each band),
- and the **magnitude of TF** (in its frequency grid).

This calibration is useful so that the RTA/TF doesn't "lie to you" about the microphone signature.

File format (practical)

1. Two columns: freq_hz and corr_db
2. It can be separated by spaces or by a comma (typical CSV/TXT).
3. The correct (conceptual) meaning is:
    a. If the microphone measures "too much" at 10 kHz, the correction is usually negative there;
    b. If it measures "less," the correction is positive.

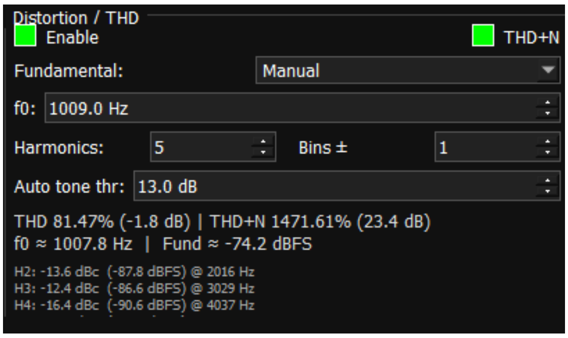
Procedure

- Open **RTA Pro Utilities → Calibration** .
- Click **Load…** and select the calibration file.
- Activá **Enable**.
- Verify that the file name appears in the label.
- If you want to return to "no calibration", use **Clear** (and/or disable Enable).

When is it appropriate to use it?

- Fine-tuning of "tonal balance" with RTA (PA, control rooms, etc.).
- Comparison against targets (House/tilt/flat) with less microphone bias.
- TF magnitude “more realistic”.

2.12. Ventana “Distortion / THD”



This module is an FFT-based distortion meter, **primarily intended for measurement with a** (sinusoidal) tone.

2. 12.1 Concepts: THD vs THD+N

- **THD (Total Harmonic Distortion)** : measures the energy of the **harmonics** (2f0, 3f0, …) with respect to the fundamental f0.

$$THD = \frac{\sqrt{V_2^2 + V_3^2 + \dots}}{V_1}$$

- **THD+N** : also includes **noise** (everything that is not the fundamental), typically in the 20 Hz–20 kHz band.

In real-world measurements, THD+N is usually "larger" because noise and non-harmonic energy are taken into account.

2.12.2 Module controls (what each one does)

***Enable:* Activates the calculation. If it is off, the panel displays “THD: —”** .

THD+N (checkbox): When activated, in addition to harmonic THD, it displays THD+N.

Fundamental (selector)

- **Auto (FFT peak):** detects f0 as the dominant peak of the spectrum (if it is sufficiently tonal).
- **Generator sine:** takes f0 from the generator frequency if it is generating a sinusoid (the most stable).
- **Manual:** f0 is defined by the user.

f0 (spinbox): Tone frequency when the mode is in “Manual”.

Harmonics: Number of harmonics to include (minimum 2, maximum 10 in this panel).

- If you put 5, they are considered 2f0..5f0.
- If the system is already close to the Nyquist or f0 is high, the system clips harmonics that do not enter the band.

Bins ±: Each harmonic is integrated into a “window” of FFT bins around the main bin.

- `Bins ± 0` : an exact bin (very sensitive to leakage)
- `Bins ±1` or `±2` : more robust (captures energy from around)

Auto tone thr: Auto only: requires that the “tone peak” be at least X dB above the “average content” of the spectrum, to avoid false positives with noise/music.

2.12.3 What it shows and how to interpret it

The panel displays:

- **THD %** and also in relative dB (a way of visualizing it as a ratio).
- If `THD+N` is active: also **THD+N %** .
- `f0 ≈ … Hz` and `Fund ≈ … dBFS` (fundamental level).

Also, list of harmonics:

- `H2, H3, …` in dBc (relative to the fundamental) and dBFS.

2.12.4 Recommended measurement procedure (practical and realistic)

1. Set the generator to **sinusoid** (e.g., 1 kHz) and power the system/device to be measured.
2. Select `Fundamental: Generator sine` to set f0 (this prevents Auto from "jumping").
3. Adjust the level to:
    a. avoid clipping at output and input,
    b. and have sufficient SNR.
4. Choose:
    a. `Harmonics` : 5 is usually a good starting point
    b. `Bins ±` : 1 or 2 for robustness
5. Note:
    a. dominant harmonics (typically H2/H3),
    b. and THD+N if the environment is noisy.

2.12.5 Limitations (important)

1. THD by FFT is sensitive to:

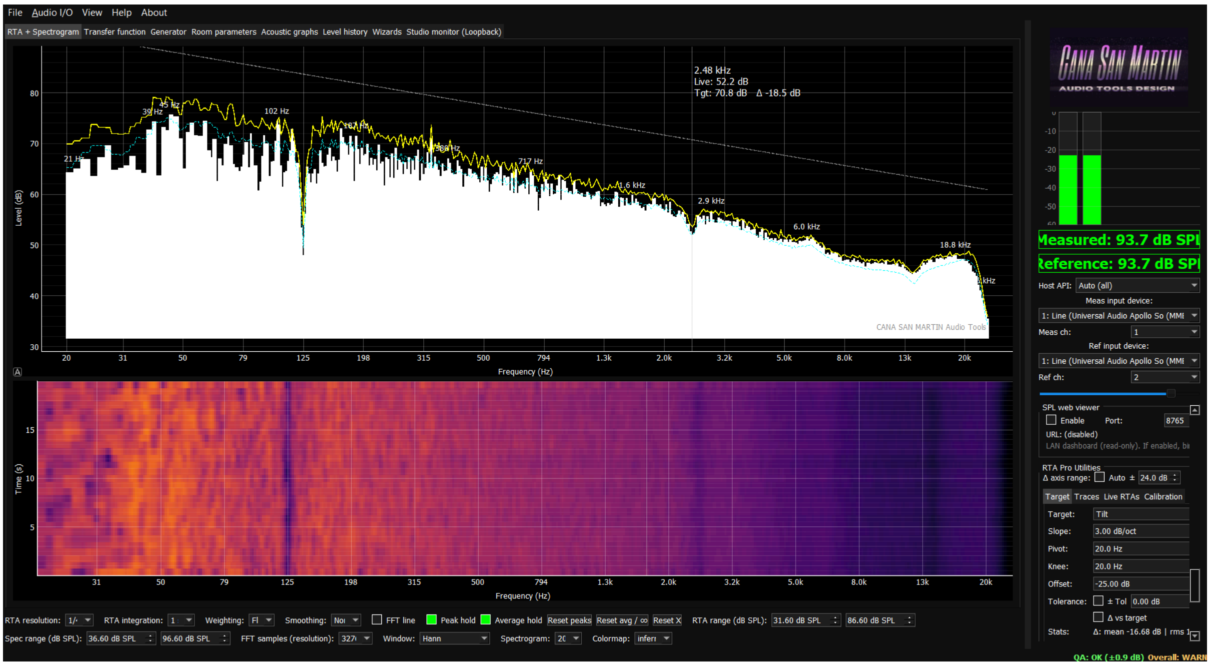
    a. FFT size / resolution,
    b. window,
    c. leakage,
    d. background noise,
    e. dynamic nonlinearities.

    With music or complex signals, **THD/THD+N ceases to make sense** as a single metric; the panel is designed for tones.

3. RTA + Spectrogram tab (real-time spectral analysis)

This tab integrates **two complementary visualizations** of the signal's spectral content:

- **RTA (Real-Time Analyzer)** : an "instantaneous" (or averaged) spectrum displaying **level versus frequency** in dB. The frequency axis is presented in **bands** (fractional octave bands) and may include an **FFT-type trace** (line) for fine detail. The RTA is intended to be the "classic" reading for quick diagnosis and tuning. Unlike other RTAs, hovering over a frequency band displays the intensity and its respective deltas, and for each octave band, it displays the highest frequency band (10 frequencies highlighted in total).
- **Spectrogram** : A time-frequency (STFT) map showing **how the spectrum evolves over time** , using a **color code** to represent the level in dB. In **SMartin** , the spectrogram is labeled with **Frequency (Hz) on the bottom axis** and **Time (s) on the left axis** , and also displays the frequency corresponding to the cursor movement .



3.1 RTA: what is seen and how to interpret it

3.1.1 What does it physically represent?

The RTA is based on time-windowed Fourier analysis (STFT): an audio block is taken, a **window is applied, an FFT** is calculated , the modulus is converted to dB and then (depending on the configuration) it is grouped by **fractional bands and/or an FFT line** is drawn .

- In band mode, the X axis is interpreted as **fractional octave bands** (1/3, 1/6, 1/12, 1/24, 1/48).
- In “FFT line” mode, the graph can display much finer resolution detail (dependent on the FFT size), useful for narrow resonances or tones.

In the widget itself, the concept is documented as follows: **“x: band index labeled in Hz; y: dB; bar mode or FFT-style line”** .

3.1.2 Important conceptual warning (for “scientific” reading)

An RTA **is not a transfer function** . What it displays is the **spectral content of the input signal** to the analyzer: it includes the sum of source, system, room, ambient noise, microphone response, and gain.
To approximate a system "response," a known excitation (e.g., **pink noise) and controlled conditions are typically used, or a transfer function** measurement is taken directly .

3.2. RTA parameters (controls)

The main RTA controls are on the bottom strip (row 1), and include: **RTA resolution** , **RTA integration** , **Weighting** , **Smoothing** , **FFT line** , **Peak hold** , **Average hold , Reset** buttons and **RTA range (dBFS)** adjustment .

3.2.1 RTA resolution (fractional-eighth resolution)

In **SMartin** , the selector offers: **1/3, 1/6, 1/12, 1/24, 1/32 and 1/48** .

Theory:

- **A 1/N octave** band divides each octave into N bands of equal logarithmic ratio.
- If f_c is the center frequency of the band, its typical edges are modeled as:

$$f_{low} = f_c \cdot 2^{-1/(2N)}, \quad f_{high} = f_c \cdot 2^{+1/(2N)}$$

- At higher N (for example 1/48), the bands are narrower: you see more detail, but also more variation due to interference (reflection combs), noise, and micro-movements.

Recommended practice/choice:

- **1/3** : Macro view (general trend). Useful for "quick check" and tonal balance.
- **1/6 – 1/12** : very common in system tuning (PA/monitors) because it balances detail and stability.
- **1/24 – 1/48** : fine diagnosis (modes, narrow resonances, localized problems), but requires more care to avoid “over-interpreting” comb filtering.

3.2.2 RTA integration (temporary integration)

Options: **Inst., 1 s, 2 s, 5 s, 10 s, ∞** .

- It is a **time average** (typically exponential or cumulative) that reduces variance of the spectral estimator.
- “Inst.” behaves like almost no average reading: very reactive, but more “nervous”.
- Values in seconds are equivalent to a time constant: the spectrum converges more slowly but is more stable.
- “∞” represents a “long-term” cumulative average (until it is reset).

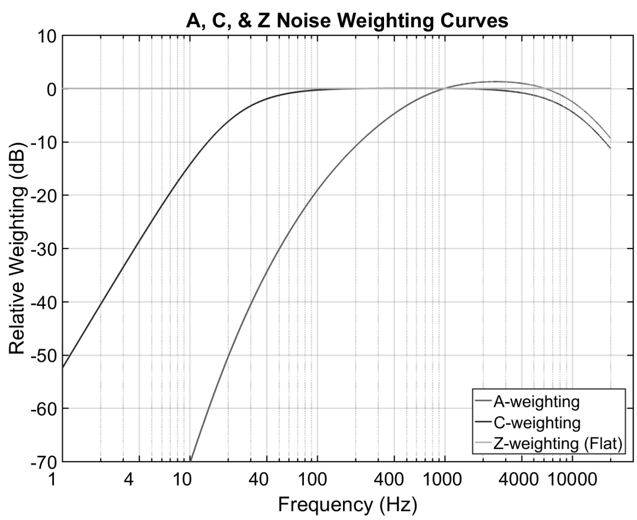
Practical for:

- **Quick search** for peaks or events: *Inst.* or 1 s.
- **Stable measurement** with pink noise: 2–10 s.
- **comparisons** (before/after) with maximum stability: ∞, resetting when the condition changes.

3.2.3 Weighting (Flat / A / C)

Options: **Flat, A,** C.

- The A and C weightings are normalized curves (IEC family) that approximate hearing sensitivity under different conditions.
- **A-weighting** attenuates bass and enhances mid-high frequencies: it is widely used in ambient noise, exposure, and “perceptual” measurement.
- **C-weighting** is flatter (especially at high levels) and is used when a less skewed measure of severe symptoms than A is desired.



Utilization:

- **Flat** : preferred for technical diagnosis of actual spectral response and content.
- **A** : preferred if the question is "how it is perceived" or for environments/regulations (according to applicable standard).
- **C** : useful in high-level measurements or when it is important to preserve more bass information than with A.

3.2.4 Smoothing (spectral smoothing)

Options: **None, 1/3, 1/6, 1/12, 1/24, 1/48** .

- Smoothing is a **post-processing on the frequency** (not temporal) axis: it reduces fast "ripple" and helps to visualize trends.

- It is not the same as “RTA resolution”: one defines bands, the other smooths the curve/values presented.

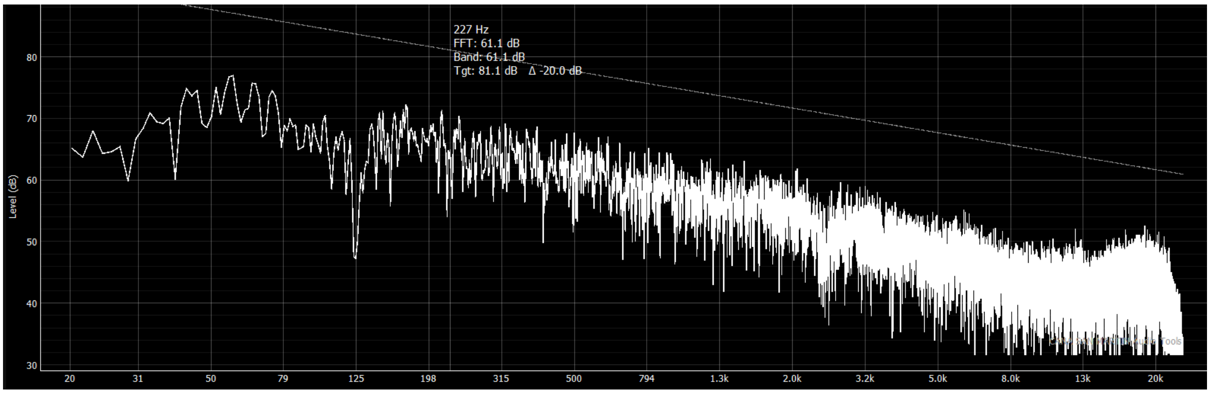
In practice:

- **None** : maximum fidelity (and maximum sensitivity to combs/reflections).
- **1/12 – 1/6** : very useful for interpretive reading in rooms (where comb filtering can be distracting).
- Avoid excessive smoothing if you are looking for tight resonances.

3.2.5 FFT line

Checkbox **FFT line** (enabled by default)

- Enables a trace that represents the spectrum with bin resolution FFT (dependent on).
- It is useful for detecting very narrow tones or peaks that can be "averaged" or "hidden" in bands.
- Activate it for **hum, whine, resonances** .
- Turn it off if you want a cleaner "band" reading comparable to fractional standards.



3.2.6 Peak hold

Checkbox **Peak hold** (enabled by default)

- It retains, by frequency/band, the **maximum** observed since the last reset. It is a "worst case" estimator for intermittent events.
- Excellent for capturing fleeting peaks (knocks, "S" in voice, resonance shots).

3.2.7 Average hold

Checkbox **Average hold** (enabled by default)

- It maintains an average (depending on implementation and time integration) that allows you to see a stable curve.
- Useful for comparing “trend” vs “peaks”, especially with pink noise.

3.2.8 Botones Reset (peaks, average/∞, X zoom)

- **Reset peaks** and **Reset avg / ∞** exist as dedicated buttons.
- Internally, the peak reset calls the widget's reset method, and the average reset clears both the normal average and the infinity average (if it exists).
- **Reset X** returns the X-axis zoom to its initial state (provided as "Reset X zoom").
- Use **Reset peaks** before a new recording so that the peak hold is comparable.
- Use **Reset avg / ∞** when you change mic position, change target, or change excitation condition.
- Use **Reset X** after zoom inspections.

3.2.9 RTA range (dBFS) (vertical range)

It is controlled by two values (minimum and maximum). Default: **−90.0 to +10.0 dBFS** .

- It's a **display decision** : compress or expand the Y-axis.
- In dBFS, 0 dBFS represents the full scale digital maximum. More negative values are lower levels.

Relationship with SPL (when applicable):
The code itself considers that what the user sees can be dBFS or dB SPL by applying an offset (calibration); internally it is reconverted to "base dBFS" by subtracting the offset before setting the plot range.

- For normal signals: a range like **−90 to −20 dBFS** can be more informative if you want to see detail in still material.
- For strong signals: zoom in upwards.
- If you saturate (clipping), the range doesn't "fix" it; it only hides/makes it visible.

3. Spectrogram: what it shows and how to interpret it

3.3.1 What does it physically represent?

The spectrogram is a time-domain STFT: for each time block, a spectrum is calculated and a history is stacked. In **SMartin** , the panel is labeled with:

- **Eje inferior:** Frequency (Hz)
- **Left axis:** Time (s)

In addition, the spectrogram includes a frequency indicator under the cursor (reading of "Hz" according to cursor position).

- **Persistent horizontal zones** (over time) indicate tonal/stable components at that frequency.
- **Broadband energy** is seen as a more distributed colored "layer".
- **Transients** appear as localized changes in the time dimension.

3.4 Spectrogram Parameters (Controls)

The spectrogram controls are in the bottom strip (row 2): **Spec range** , **FFT samples** , **Window** , **Spectrogram (duration)** and **Colormap** .

3.4.1 Spec range (dBFS) (color range / contrast)

Default: **−80.0 to −20.0 dBFS** .

- Define the mapping: what level is considered “minimum” (darker color) and what level is considered “maximum” (more intense color).
- It's equivalent to selecting the **dynamic** heatmap window. It doesn't change the calculation, it changes the visibility.
- If the spectrogram looks "all the same", narrow down the range.
- If everything is "saturated" (too bright), increase the maximum or decrease the gain.

SPL Note:
The same SPL offset criterion is also considered: the visible range can be in dBFS or with offset, but it is translated to dBFS when applying the actual range.

3.4.2 FFT samples (resolution) (FFT size)

Options: **2048, 4096, 8192, 16384** .

1. Frequency resolution:
   $\Delta f = \frac{f_s}{N}$

   where f_s is samplerate and is the FFT size.
2. Temporal resolution (roughly speaking) is given by the duration of the window:
   $T \approx \frac{N}{f_s}$
3. Increase N:
   a. Improves frequency detail (finer).
   b. It worsens temporary reactivity (making it “slower”).
   c. It increases computational costs.

Example (at 48 kHz):

- N=2048 → Δf≈23.4 Hz; ventana≈42.7 ms
- N=8192 → Δf≈5.86 Hz; ventana≈170.7 ms
- N=16384 → Δf≈2.93 Hz; ventana≈341 ms

In practice:

- For **bass/modes** and fine tones: Large FFT (8192–16384).

- For **transients** or fast reading: Lower FFT (2048–4096).

3.4.3 Window (analysis window)

Options: **Hann, Hamming, Blackman, Rectangular** .

The window is the weight applied to the temporary block before the FFT to control:

- **Leakage** (spectral leak, energy "spilled" into neighboring bins).
- **Main lobe width** (effective resolution).
- **Level of lateral lobes** (leakage artifacts).

Typical comparison:

- **Rectangular** : maximum nominal resolution, worst leakage (high side lobes).
- **Hann** : excellent balance; widely used as a default.
- **Hamming** : similar to Hann, with different involvement of lobes.
- **Blackman** : lower side lobes (less leakage), but wider main lobe (less "sharpness").

In practice:

- Recommended default: **Hann** (robust for general use).
- If you have strong tones and want to minimize "tails": **Blackman** .
- Avoid **rectangular** unless you know why you need it (leakage can be confusing).

3.4.4 Spectrogram (duration / visible history)

Options: **10s, 20s, 50s** .

1. Controls how many seconds of history are displayed and buffered.
2. The longer the duration:
    a. More context (slow events).
    b. Lower visual "density" per second if the panel height is fixed.
    c. Higher memory/CPU usage (depending on implementation).
3. 10 s: rapid diagnosis, transients.
4. 20s: general equilibrium.
5. 50s: slow evolution (intermittent noises, drift, gradual changes).

3.4.5 Colormap (color palette)

Options: **inferno, magma, plasma, viridis, gray** .

- The data doesn't change; the visual mapping function changes.
- "Perceptually uniform" palettes (e.g., viridis) help to interpret level differences without bias.
- **inferno/magma/plasma** : high contrast on dark backgrounds.

- **viridis** : good for general readability and perceptual consistency.
- **gray** : useful for monochrome printing or documentation.

3. 5) Practical recommendations (for acoustic measurement and diagnosis)

1. **First, choose the measurement question**
   a. "What's playing?" → RTA + Spectrogram (real material).
   b. "How does the system respond?" → controlled stimulus + (ideally) transfer function.
2. **For "response" with RTA**
   a. Recommended excitation: **pink noise** (constant octave energy).
   b. RTA resolution: 1/6–1/12
   c. Integration: 2–10 s (o ∞ con reset)
   d. Smoothing: None or 1/12 (depending on the objective)
3. **For tonal problems** (hum, hissing, narrow resonances)
   a. FFT line ON
   b. FFT samples altos
   c. Window Hann/Blackman
   d. Spectrogram 20–50 s
4. **Manage scales (range)**
   a. Adjust **RTA range** and **Spec range** to avoid "squashing" the contrast.
   b. Reset peaks/averages before comparing.

4.1 Transfer Function Tab

The **Transfer Function (TF) tab** is designed to **analyze the relationship between a reference signal and a measured signal** in the frequency domain. Its main purpose is:

- Align systems (time, phase, and magnitude).
- Verify the actual frequency response of the electroacoustic system.
- Evaluate the consistency and reliability of the measurement.
- Compare different systems, channels, fills, subs, delays, or configurations.

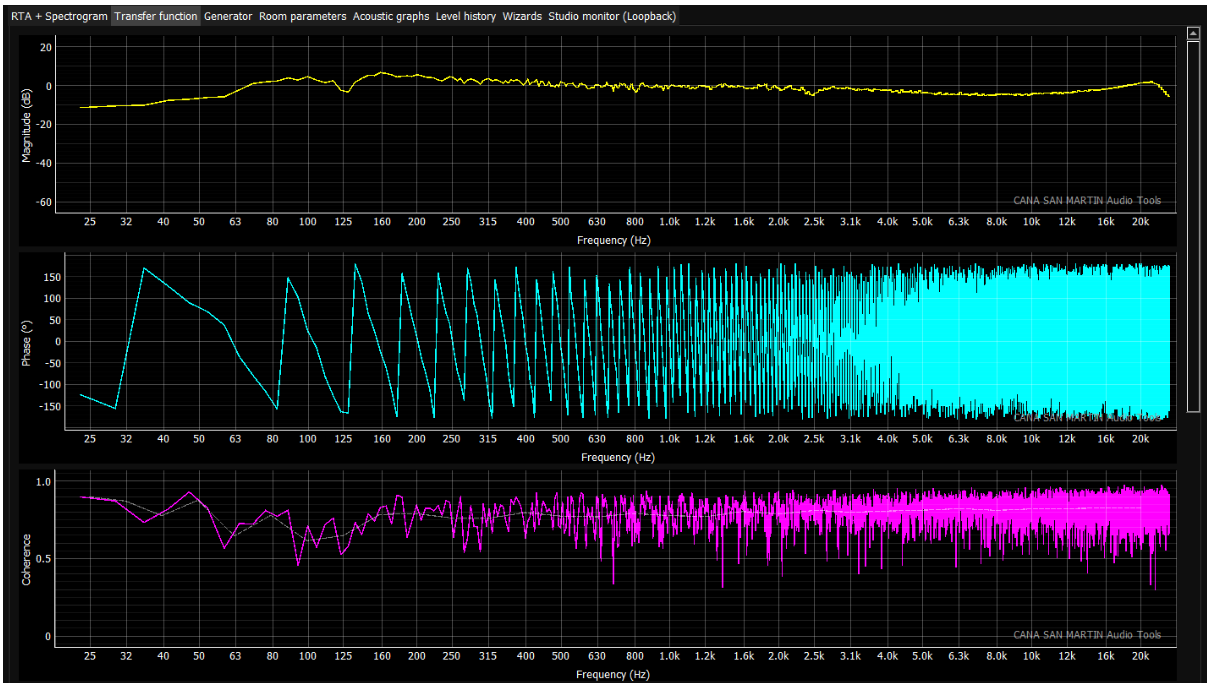
From a mathematical point of view, the transfer function is defined as:

$$H(f) = \frac{Y(f)}{X(f)}$$

where:

- $X(f)$ It is the FFT of the **reference signal** .
- $Y(f)$ It is the FFT of the **measured signal** .

- $H(f)$ Describe how the system modifies amplitude and phase as a function of frequency.



4.2 . General architecture of the tab

The tab is organized into three conceptual blocks:

1. **TF Engine Management (TF Engines / Channel Mapping)**
2. **Analysis graphs (Magnitude, Phase, Coherence)**
3. **Controls and parameters for processing and display**

All TF analysis shown depends **exclusively on the active TF engine** .

4.3. TF Engines / Channel Mapping

4.3.1 What is a TF Engine

A **TF Engine** defines **which channels** are used for:

- Measured signal (*Measurement channel*).
- Reference signal (*Reference channel* or virtual reference).

Each motor is **independent** , allowing multiple simultaneous TF cards. Channels are numbered from 0 (0 = ch 1; 1 = ch 2).

4.3.2 Parameters of each engine

For each TF Engine, the following is defined:

1. **Enabled**
   Activates or deactivates the calculation of that engine.
2. **Name**
   Logical name of the engine (e.g., *Main* , *Subs* , *Frontfills*).
3. **Meas ch**
   Physical input channel corresponding to the microphone or measured point.
4. **Ref source**

   a. *Input* : reference taken from another input channel.
   b. *Generator* : internal reference from the generator.

5. **Ref ch**
   Reference channel (typically console output or loopback).
6. **Delay (s)**
   Delay applied **only to the reference** for fine time alignment.
7. **Overlay**
   allows you to overlay the TF of that engine on top of the main TF.
8. **Color:**
   The color assigned to visually identify that engine.

4.3.3 Practical use

- **Main** Motor : Main PA measurement.
- Motor **Subs** : dedicated microphone for subwoofers.
- Motor **Fill** : measurement of frontfills or delays.
- Direct comparison between systems without changing routing.

4. Magnitude Graph

4. 4.1 What does it show

The Magnitude graph represents:

$$|H(f)| \quad en\, dB$$

In other words, **how the system amplifies or attenuates each frequency** relative to the reference.

4.4.2 Interpretation

- Slopes → tonal balance of the system.
- Steep valleys → cancellations (acoustic or electrical).
- Narrow peaks → resonances.

- Differences between engines → differences in coverage or alignment.

4.4.3 Target Curves

The magnitude can be superimposed on **target curves** , used as a reference for adjustment.

Available types:

- **Off** : sin curva.
- **Flat** : flat response.
- **HF Tilt** : progressive tilting at high frequencies.
- **House / House curve** : typical curve of a sound reinforcement system.
- **Custom CSV** : curve imported from an external file.

Associated parameters:

- **Slope (dB/oct)**
  Curve inclination.
- **Pivot (Hz)**
  Reference frequency where the curve crosses 0 dB.
- **Knee (Hz)**
  Point where the slope changes (more realistic curves).
- **Offset (dB)**
  Vertical displacement of the curve.
- **Tolerance (dB)**
  Tolerance band around the target.
- **Show tolerance**
  Show upper and lower limits.
- **Delta**
  shows the difference between measurement and target.

Typical use:
Adjust the system so that the actual TF is **within the** target tolerance.

4.5 . Phase Diagram

4.5.1 What it shows

The **phase** represents the $H(f)$ angle in degrees:

$$\angle H(f)$$

Describe **the relative frequency delay** between reference and measurement.

4.5.2 Interpretation

- Linear phase → coherent and well-aligned system.
- Constant slope → fixed delay.
- Abrupt changes → filters, crossings, reflections.
- Erratic phase → unreliable measurement (see consistency).

4.5.3 Practical use

- Alignment between subs and tops.
- Fine-tuning delays.
- Polarity verification.
- Evaluation of FIR/IIR filters.

4. 6. Coherence Graph

4.6.1 What is coherence

Coherence ($\gamma^2(f)$) measures **how correlated** reference and measurement are:

$$0 \leq \gamma^2(f) \leq 1$$

1 → highly reliable measurement.
0 → measurement dominated by noise or unrelated signals.

4.6.2 Interpretation

Low values may be due to:

- Ambient noise.
- Saturation.
- Poor SNR ratio.
- Belated reflections.
- Routing error.

4.6.3 Coherence Band Overlay

SMartin allows you to define a **coherence band per octave fraction** , which smooths the reading and helps to evaluate reliable areas in a more musical way.

Parameter:

- **Coherence band fraction** (ej. 1/3, 1/6, 1/12 oct)

4.7 . Processing Controls

4.7.1 Smoothing

Applies spectral smoothing to Magnitude and Phase.

- Fractions of an octave (e.g., 1/48 by default).
- Reduces visual noise without altering the base calculation.
- Essential for reading TF music.

4.7.2 Averaging / Integration

Time averaging of FFTs:

- Stabilizes the TF.
- Reduces instantaneous variability.
- Improves visual consistency.

4.8 . Zoom and navigation

1. **X-axis zoom**
   allows focusing on specific frequency ranges without altering the Y-axis.
2. **Reset X zoom**
   Returns to full audible range.
3. **Vertical scrolling (if applicable)**
   Makes it easier to work with taller graphics.
4. **Interactive hover:**
   When you hover the cursor over the screen, the following are displayed:

   a. Exact frequency.
   b. Magnitude / phase / point coherence.

4.9. Multi-engine overlays

When multiple TF Engines are active with *Overlay* :

- Multiple TFs overlap in the graphs.
- Each engine retains its color.
- It allows direct comparison without changing the measurement.

Typical use:

- Comparar PA vs frontfills.
- Compare microphone positions.
- Evaluate uniformity of coverage.

4.10 . Relationship with other tabs

- **Room Parameters**
  The active TF can be stored for acoustic calculations.
- **Acoustic Graphs**
  Graphs derived from TF (EDC, ETC) can be generated.
- **Wizards**
  Alignment Assistants directly use TF data.

4.11 . Good TF measurement practices

- Ensure **a clean and direct reference** .
- Check consistency before interpreting magnitude/phase.
- Do not "correct" areas of low coherence.
- Align first in time, then in phase, then in magnitude.
- Use targets as **a guide** , not as dogma.

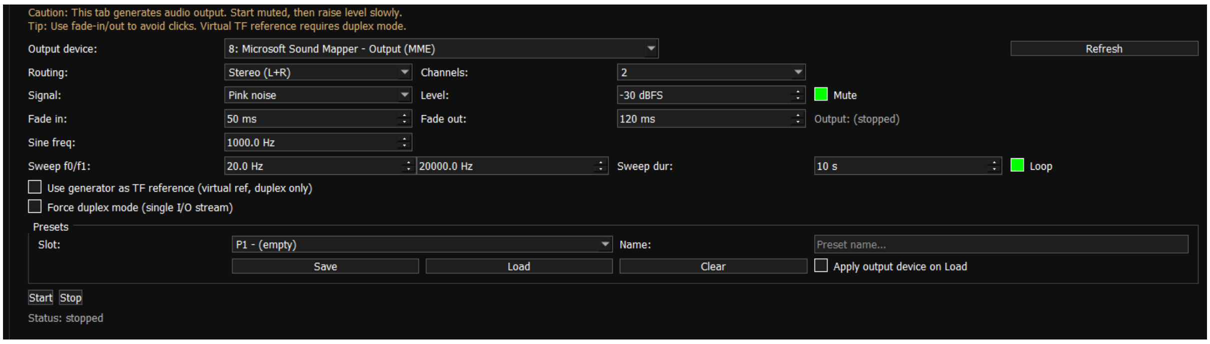
5. Generator Tab

Test signal generator for acoustic and electroacoustic measurement

The **Generator tab** provides an internal signal generator specifically designed for **measuring, aligning, calibrating, and verifying audio systems** . Its purpose is not artistic or creative, but **metrological** : to generate controlled, repeatable, and calibratable signals.

These signals can be used for:

- **Transfer Function** Measurement
- **RTA** Adjustment
- Temporal alignment (delays)
- Polarity verification
- Filter and equalization adjustment
- Room and system response measurement



5.1 . Output Device

Selector for the **output audio device** used by the generator.

Define **which board/interface** the test signal is emitted through.

- In professional measurement systems, it must match the **physical output** that feeds the system under test (PA, monitor, amplifier, processor).
- In studios, it can be routed to monitors, console buses, or virtual outputs.

Technical considerations

- The generator works in **dBFS** , so the actual level will depend on the subsequent calibration of the system.
- The device's sampling frequency directly impacts the resolution of sweeps and tones.

5.2. Output Channel / Routing (Canal de salida)

Selector of the **physical or logical channel** of the output device.

Allows you to send the signal to:

- Left channel (L)
- Right channel (R)
- Both channels (L+R)
- Specific channels in multi-channel interfaces

Practical use

- **Sub alignment** : send signal only to the channel that feeds subs.
- **Pathway comparison** : excite one pathway at a time.
- **Mono measurements** : avoid unnecessary stereo excitation.

5.3 . Generator Type / Signal Type

The generator offers different types of signal, each with a specific purpose.

5.3.1 Sine (Sinusoidal wave)

Pure signal of a single frequency.

Main use

- Polarity verification
- Delay alignment
- Identification of resonances
- Fine-tuning of phase

Advantages

- Maximum precision in time and phase.
- Ideal for observation in Transfer Function.

5.3.2 Pink Noise

Noise with **constant energy per octave** .

Main use

- Equalization adjustment
- Measurement with RTA

- Tonal balance adjustment

Advantages

- It better represents how we perceive sound.
- Standard in acoustics and sound reinforcement.

5.3.3 White Noise

Noise with **constant energy per Hz** .

Main use

- High-resolution systems analysis
- DSP technical tests

Warning

It can be **very harsh** acoustically. Use with caution.

5.3.4 Sweep / Chirp (Frequency Sweep)

A signal that traverses a range of frequencies over time.

Main use

- Impulse response measurement
- Room analysis
- Calculation of acoustic parameters

5. 4. Frequency *(for tones)*

Fundamental frequency of the sinusoidal tone, in Hz.

Typical use

- 1 kHz: standard reference
- 60–80 Hz: subwoofer alignment
- 100–250 Hz: critical crossover region
- 1–4 kHz: maximum hearing sensitivity

5. 5. Level (Output level, dBFS)

Digital level of the generator, expressed in **dBFS** . Controls the **amplitude of the digital signal** before any DA conversion.

Correct use

- Always start at low levels (e.g., -30 dBFS).
- Increase gradually while observing VU, SPL and coherence.
- Avoid digital clipping and analog saturation.

Relationship with calibration

This level only becomes **actual dB SPL** after:

- Calibrate the system
- To know the total profit of the chain

5.6. Mute / Start / Stop

Generator activation control.

- **Start** : Starts continuous broadcast.
- **Stop / Mute** : silences the output without altering parameters.

Good practices

- Always mute before changing routing or device.
- Avoid accidental activation at high levels.

5.7 . Generator Integration with other tabs

The generator is designed to work **in conjunction** with:

- **Transfer Function** : coherent excitation for magnitude, phase and coherence measurement.
- **RTA** : continuous excitation for spectral analysis.
- **Room Parameters** : Signal generation for impulse response.
- **Studio Monitor** : Full digital chain verification.

5.8 Professional considerations

- **The SMartin** generator **does not replace** an external laboratory source, but it is optimized for **field measurement and system tuning** .
- The priority is **stability, repeatability, and control** , not creativity.
- The entire generator design assumes a professional workflow: **measure** → **adjust** → **verify** .

6. Room Parameters Tab

The **ROOM PARAMETERS tab** is designed to acoustically characterize a room from an **impulse response (IR)** , showing its time representations (IR, **ETC** , **EDC/Schroeder**) and calculating classic parameters (EDT, RT by sections, clarity, definition, Ts, DRR, ITDG) with practical comparison tools **(A/B snapshots)** , **multi-position averaging** and **export** .

You obtain an **IR** in two ways:

1. **Capturing** the “current” IR from the **Transfer Function tab** (*Capture from TF button*), or
2. **Measuring** an IR with a **logarithmic sweep + deconvolution (** *Measure IR (Sweep)* group).

- Visualize:

    1. **Impulse response** (IR)
    2. **EDC (Energy Decay Curve)** by Schroeder integration
    3. **ETC (Energy Time Curve)**

- Calculates and reports **broadband parameters** : **EDT, T10/T15/T20/T30, Clarity (C50/C80), Definition (D50), Ts, DRR, ITDG + QA** (reliability indicators).
- Optionally, it performs **band analysis** (1/1 or 1/3 octave) with graphs and export to CSV.
- It allows **Store & Compare** with **A/B snapshots (overlays and differences) and a multi-position average** stream (average of several captures).

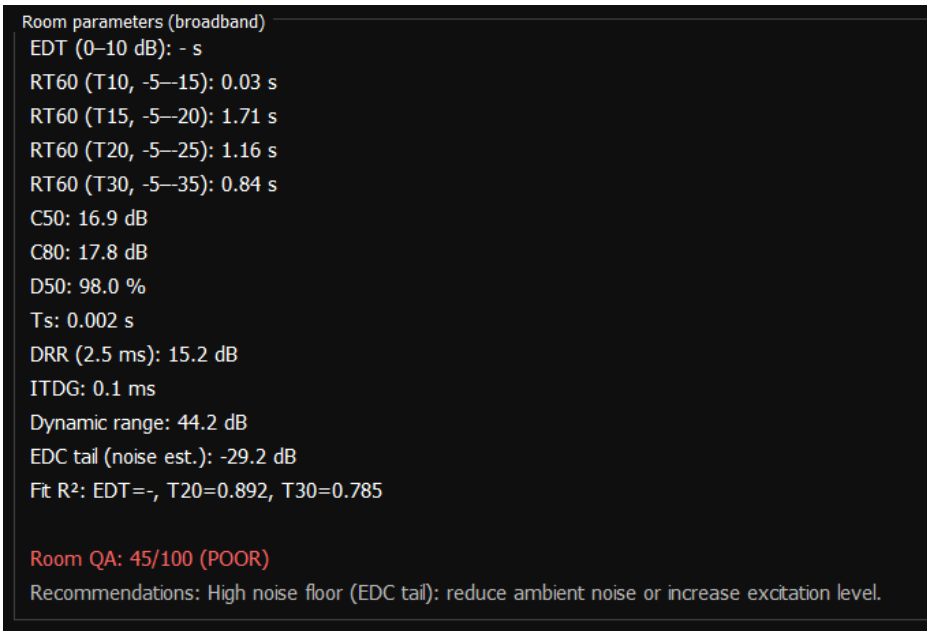
6.2 How to obtain the IR (module input)

6.2.1 Captura desde Transfer Function (“Capture from TF”)

- The program triggers a specific handler when you press *Capture from TF* , which invokes `room_widget.capture_from_transfer_function()` to bring the IR from Transfer Function and use it as the basis for computation in Room Parameters.
- If IR is not available, functions such as export will explicitly report it (“Capture from TF first”).

Typical use: when you are already measuring TF (magnitude/phase) and want to derive “room stats” from that same measurement without running a separate sweep.

WARNING: In this type of methodology, if noise levels are not optimal, the results are usually erroneous. It is essential that the warning light is green for the accuracy of this type of analysis, without exception.



6.2.2 Measurement with sweep (Measure IR (Sweep) group)

Sweep controls

The UI displays sweep and playback/capture parameters:

- **Start (Hz)** / **End (Hz)** : sweep frequency limits.
- **Duration (s)** : sweep duration.
- **Level (dBFS)** : Sweep output level in dBFS (digital level).
- **Pre-silence (s)** : time of silence before the sweep.
- **Post-tail (s)** : capture tail after the sweep to record late decay.
- **Output** : output routing (channel/device according to internal enumeration).
- Measure IR button **(Sweep)** + status label (**“Idle”** / progress).

Theory/fitting criteria (why these parameters exist)

1. **Start/End (Hz)** defines the excited spectral content. If the goal is RT/broadband clarity, it is advisable to cover the system's range of interest.
2. **Duration (s)** controls SNR and robustness at low frequencies: longer sweeps usually improve the effective SNR (at the cost of time and possible sensitivity to changes).
3. **Pre-silence** and **post-tail** are critical for:
    a. estimate background noise before excitation, and
    b. sufficiently capture the tail for RT and parameters that depend on decay.
4. **Level (dBFS)** is the digital "drive". In practice, it is adjusted to achieve good SNR **without clipping** at any stage (D/A, amplification, speaker, A/D).

What it does internally:

The sweep flow includes generating the sweep and then obtaining the IR signal through **frequency domain deconvolution** (FFT, spectral division by the sweep, IFFT to return to IR). This is explicitly stated in the sweep worker code, including the IR signal assembly via spectral operation and return to the time domain.

6.3. Temporary window controls, alignment, and bookmarks

These controls define **how IR is “cropped” and temporally referenced** , and therefore directly impact clarity/definition/DRR/ITDG and (to a lesser extent) RT settings if the cropping is aggressive.

6.3.1 IR length (s)

- Control: **IR length (s)** (spin).
- the_smartest_rta_v5_2_147
- Defines the length of the IR segment considered (the main analysis “window”).

Practical interpretation:

- Very short: you can truncate decay → RT/EDT and associated metrics become biased.
- Very long: you incorporate more background noise (noise floor dominated tail) → RT settings become less reliable and QA worsens.

Default value (restoration): 4.0 s.

6.3.2 Align to peak

- Control: checkbox **Align to peak**.
- Concept: align t=0 to the **main peak** (usually the direct arrival), so that “Early/Clarity/DRR” measure from a consistent reference.

When it is convenient: practically whenever you want to compare measurements or extract "early" parameters (clarity, D50, DRR, ITDG).

Default value (restoration): enabled (`True`).

6.3.3 Temporary "early" and "direct" windows

The UI displays three key times (in milliseconds):

- **Early (ms)** : "early" integration limit for clarity and definition (e.g. C50/D50 if Early=50 ms).
- **Clarity2 (ms)** : second limit for alternate clarity (typically 80 ms if used as "C80").
- **DRR (ms)** : "direct" window used to calculate DRR (direct energy vs remainder).

Default values (restoration):

- Early = 50.0 ms
- Clarity2 = 80.0 ms
- DRR = 10.0 ms

6.3.4 "Show markers" / "Edit markers"

1. Checkboxes:
   a. **Show markers**
   b. **Edit markers**

What do they mean?

- **Show markers** enables the display of markers (vertical lines/indicators) associated with IR events/windows and, in particular, can add reflection markers on the ETC.
- **Edit markers** enables manual adjustment of these markers (useful when the "direct sound" or main reflections are not detected correctly by default, or when you want to set a different "direct" window for discretionary reasons).

Default values (restoration): both disabled (False).

6.4. Visualizations (what each graph shows)

This tab contains three main plots, all with grid, clipping and horizontal navigation, and with a minimum height set for comfortable reading.

6.4.1 Impulse response (IR)

- Plot: `ir_plot` (impulse response).

What to watch:

- Main peak (direct arrival).
- Early reflections (later energy packs).
- Possible saturations (flattened shape) or excessive noise.

6.4.2 EDC (Energy Decay Curve)

- Plot: `edc_plot` , labeled as **EDC (dB)** , with typical vertical range **-80 to +5 dB** .
- The EDC is obtained by inverse cumulative integration of the energy (Schroeder) over the square of the IR. In practical terms, it allows for the estimation of reverberation times from slopes (linear regressions in dB).

6.4.3 ETC (Energy Time Curve)

1. Plot: `etc_plot` , labeled as **ETC (dB)** , with vertical range **-80 to +5 dB** .
2. It displays relative energy over time (direct + reflections), ideal for diagnosing:
    3. strong early reflections,
    4. “clusters” of reflections,
    5. windows relevant to DRR and ITDG.

6.4.4 Tabla Early reflections (ETC)

- An “Early reflections (ETC)” module is included with a 3-column table: **#** , **Time (ms)** , **Level (dB)** .

What it offers:

- Lists detected reflections (up to a top-N) and allows quantifying their relative times and levels.

6.5 Room Metrics (Broadband)

The metrics block exposes explicit labels for:

- **EDT**, **RT10**, **RT15**, **RT20**, **RT30**
- **Clarity** (dos lecturas), **Definition** , **Ts** , **DRR** , **ITDG**
- **Dynamic range**, **Noise tail**, **Fit R²**, **Room QA** y **Warnings**.

Operational note: Clarity and definition labels are dynamically rendered as **C{Early}** , **C{Clarity2}** , **D{Early}** , using the spinner values (Early ms and Clarity2 ms).

6.5.1 EDT and RT by sections (T10/T15/T20/T30 → RT60)

- **EDT** (Early Decay Time): estimates the initial decay; it is very sensitive to early reflections and the "character" of the onset of decay.
- **T10/T15/T20/T30** : reverberation time estimates using different decay sections in the EDC (in practice, different dB windows and regression).
- **RT60** : extrapolation to 60 dB (where applicable) from the measured sections.

Practical reading:

- EDT vs T20/T30: large differences usually indicate that the decay is not "monoexponential" (room with multiple regimes, high noise, irregular late energy, etc.).
- Use QA and Fit R^2 as an indicator of regression reliability.

6.5.2 Clarity: C{Early} y C{Clarity2}

The system calculates two clarity metrics:

- **C_early** (associated with **Early (ms)** , by default 50 ms → C50)
- **C_2** (associated with **Clarity2 (ms)** , by default 80 ms → C80)

Definition (the one implemented by the software):

- `Early energy E_early` is calculated by integrating `ir²` from 0 to `early_s` .
- Late energy `E_late = E_total - E_early` .
- **Clarity**:

$$C = 10log_{10}\left(\frac{E_{early}}{E_{late}}\right)$$

This is explicit in the calculation of `C_early` and `C_2` .

Practical interpretation:

- High Clarity: greater proportion of early energy (perception of greater definition/“dry”).
- Clarity low: more late energy (more “ambience”/tail).

6.5.3 Definition: D{Early} (typically D50)

Definition (implemented):

$$D = 100 \cdot \frac{E_{early}}{E_{total}}$$

where `E_early` uses the same `Early (ms)` as Clarity early. It is implemented as `D_early` .

Practical interpretation:

- It is expressed as a percentage.
- It is another (more "energetic") way of quantifying how much energy arrives early.

6.5.4 Ts (Center Time)

Definition (implemented):

$$T_s = 1000 \cdot \frac{\sum t \cdot ir(t)^2}{\sum ir(t)^2}$$

(Expressed in ms). Implemented as `Ts_ms` (center time).

Practical interpretation:

- Lower values: energy concentrated earlier.
- Higher values: greater relative weight of late energy (tail/reverberation).

6.5.5 DRR (Direct-to-Reverberant Ratio)

The DRR is calculated from a "direct" window and the rest of the energy:

1. Direct window: from 0 to `drr_s` (defined by **DRR (ms)**).
2. Direct energy `E_direct` .
3. Remaining energy `E_rest = E_total − E_direct` .
4. Definition (implemented):
5. $$DRR = 10log_{10}\left(\frac{E_{direct}}{E_{rest}}\right)$$

Implemented as `DRR_db` .

Practical interpretation:

- A higher DRR implies a greater predominance of the direct sound field over the reverberant field, which usually correlates with greater "control"/clarity in the listening position (depending on the context).

6.5.6 ITDG (Initial Time Delay Gap) and early reflections

In the software, the ITDG is defined as the **earliest significant reflection** after the live broadcast.

Algorithm implemented:

- Local maxima are sought in the ETC after the main (direct) peak, in a time window, with a threshold (for example `thr_db = -20 dB`).
- `A list refl = [(dt_ms, lvl_db), ...]` is formed , sorted, and a top-N is taken.
- **ITDG** is taken as the minimum `dt_ms` found in those reflections.

Associated output:

- The "Early reflections (ETC)" table is updated with these reflections (time and level), and optionally the system places vertical markers in the ETC when "Show markers" is activated.

Practical interpretation:

- Small ITDG: early reflections very close to the direct (potentially more coloration/comb filtering if strong).
- ITDG major: major "gap" before the first significant reflection.

6.5.7 Reliability Indicators: Dynamic range, Noise tail, Fit R^2, QA, Warnings

In addition to acoustic parameters, the module displays:

- **Dynamic range**
- **Noise tail**
- **Fit R^2**
- **Room QA**
- **Warnings**

And in band export, fields such as `dyn_range_db` , `noise_tail_db` , `fit_r2` and `warnings are explicitly included` along with the acoustic parameters (this is useful for reports and for validating measurement quality).

- **Low Fit R^2** or high **Noise tail** usually indicate that the decay section is contaminated by noise or that the decay is not linear in dB (multiple slopes).
- **Warnings** concentrates automatic observations to prevent misinterpretations (e.g., too noisy tail, insufficient dynamic range).

6.6) Band parameters: analysis in 1/1 and 1/3 octave

6.6.1 Controls

- **Band mode** (combo): `Off`, `1/1 octave`, `1/3 octave`
- **Compute bands** button
- **Export bands CSV** button

6.6.2 What the band charts show

The banding module includes dedicated visualizations, for example:

- **Decay times (per band)** : EDT curves/series **, T10, T15, T20, T30** as a function of `fc (Hz)`
- **Clarity/DRR (per band)**: curvas de **C50, C80, DRR, Ts (ms)** vs `fc (Hz)`
- **Definition (per band)**: **D50 (%)** vs `fc (Hz)`

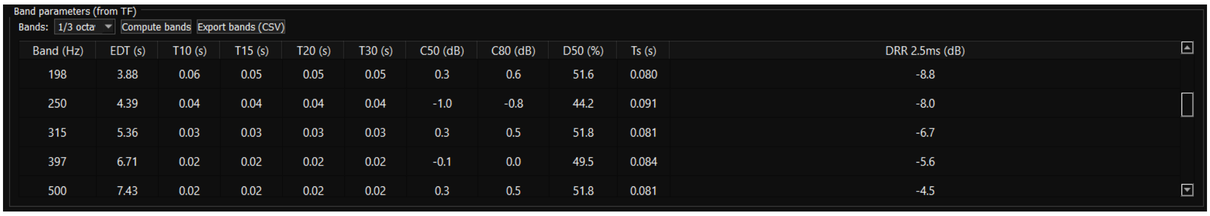
6.6.3 Theory: why by bands

The acoustic behavior of a room **is not uniform across frequencies** : long modes/times predominate in the bass frequencies; in the mid-high frequencies, absorption and diffusion typically reduce RT.
Analysis by 1/1 or 1/3 octave reveals:

- excessive resonances and tails in specific bands,
- imbalances of “clarity” or “definition” by band,
- consistency with design/use objectives (control room, recording room, etc.).

6.6.4 Export by bands (CSV)

El export incluye una cabecera con parámetros por banda (ej.: `fc_hz`, `EDT_s`, `T10_s`, `T15_s`, `T20_s`, `T30_s`, `RT60_s`, `C50_dB`, `C80_dB`, `D50_pct`, `Ts_ms`, `DRR_dB`, `ITDG_ms`, `dyn_range_db`, `noise_tail_db`, `fit_r2`, `warnings`).


Band parameters (from TF)
Bands: 1/3 octa Compute bands Export bands (CSV)

Band (Hz)	EDT (s)	T10 (s)	T15 (s)	T20 (s)	T30 (s)	C50 (dB)	C80 (dB)	D50 (%)	Ts (s)	DRR 2.5ms (dB)
198	3.88	0.06	0.05	0.05	0.05	0.3	0.6	51.6	0.080	-8.8
250	4.39	0.04	0.04	0.04	0.04	-1.0	-0.8	44.2	0.091	-8.0
315	5.36	0.03	0.03	0.03	0.03	0.3	0.5	51.8	0.081	-6.7
397	6.71	0.02	0.02	0.02	0.02	-0.1	0.0	49.5	0.084	-5.6
500	7.43	0.02	0.02	0.02	0.02	0.3	0.5	51.8	0.081	-4.5

6.7 Room snapshots: Store & Compare (A/B)

It allows you to "freeze" measurement states to compare conditions (treatment before/after, microphone/position, speaker, window settings, etc.).

6.7.1 Visible Controls

In the **Room snapshots (Store & Compare) group** :

1. Botones: **Store A**, **Store B**, **Clear**
2. Overlays: **Show IR**, **Show EDC**, **Show ETC**
3. Comparison mode selector:
    a. **A vs B**

   b. **A only**
   c. **B only**
4. Text/result area: `snap_compare_label` (comparative summary).

6.7.2 Recommended use

- **A** : baseline condition.
- **B** : modified condition.
- Activate overlays (IR/EDC/ETC) to view structural changes (direct peak, reflections, EDC slope, etc.) along with the quantitative summary.

6.8 Multi-position average

6.8.1 What it is and why it exists

The goal is to obtain metrics **representative** of a zone (e.g., listening area) rather than relying on a single point. Multi-position averaging reduces the spatial bias typical of small rooms (nodes/antinodes in bass frequencies, variations due to small displacements, etc.). The module explicitly declares this as a function of the tab (capture and average calculation).

6.8.2 Controls

In the **Multi-position average group** :

- **Target positions** (spin)
- Botones: **Capture position**, **Clear**, **Compute avg**
- Buttons: **Store avg → A** , **Store avg → B**
- Indicators: **Captured: n** and a label **Avg:** (average status).

Default values (restoration): Target positions = 5.

6.8.3 Recommended practice

- Define a small grid around the listening point (or representative points of the area).
- Capture positions with consistent conditions (same speaker, same level, same IR window).
- Calculate the average and, if the goal is comparison, store that average in A or B and compare against another condition.

6.9 Exports (for reporting and reuse)

6.9.1 Export IR (WAV)

- Button: **Export IR (WAV)**
- Operational requirement: A captured/loaded IR must exist; otherwise, "No impulse response to export. Capture from TF first." will be reported.

Typical use:

- Archive measurements,
- work offline in another environment,
- document "before/after" with exportable IR.

6.9.2 Export bands CSV

- Button: **Export bands CSV**
- Includes band and QA parameters in columns (see header).

6.10 Quick Start Guide (Recommended Procedure)

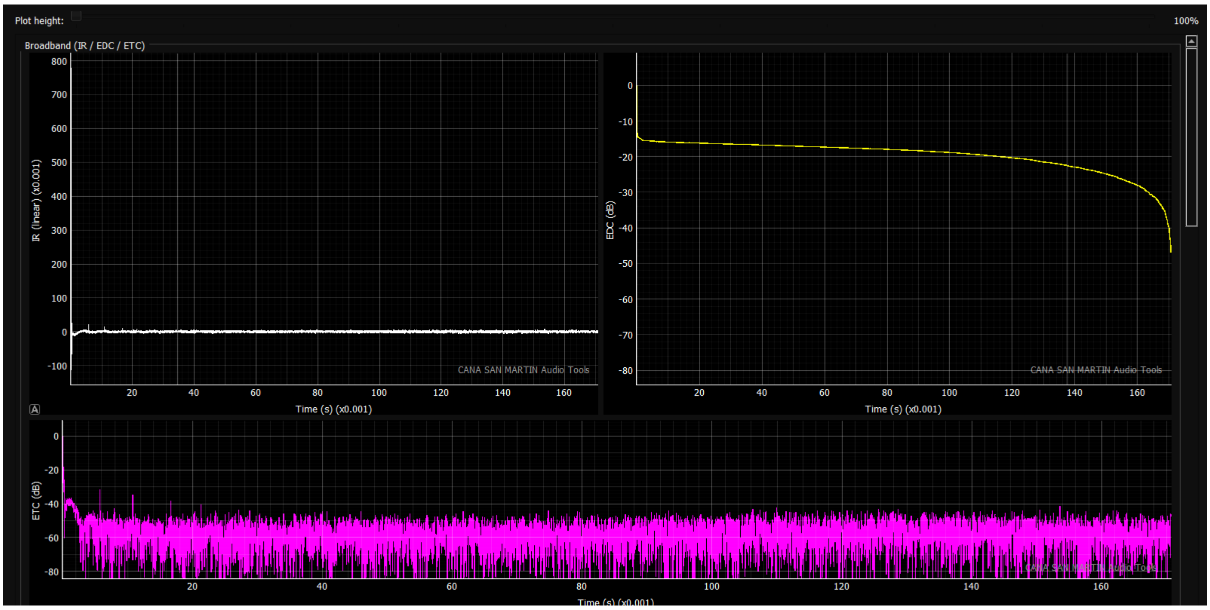
- **Preparation**
- Choose the **speaker** to measure (L/R/mono), disable unwanted processing (limiters, automix, etc.).
- Place the **microphone** in the target position (e.g., listening point). Ideally, use a measurement microphone with consistent orientation.
- **Get IR**
- Option A: Measure Transfer Function and then **Capture from TF** .
- Option B: measure with **Measure IR (Sweep)** adjusting Start/End, Duration, Level, Pre-silence, Post-tail and Output.
- **Adjust references and windows**
- Check **Align to peak** and adjust **IR length** to include useful decay without "eating up" too much noise floor.
- Define Early/Clarity2/DRR according to the type of analysis (speech/music/control room, etc.).
- **Interpret plots**
- Use **ETC** for early reflections/ITDG and reflection table.
- Use **EDC** for decay consistency and RT.
- **Comparison / average**
- Save **A/B** and compare overlays and summary.
- If spatial representativeness is needed, use **Multi-position average** with target and compute avg.
- **Bands and export**
- Activate 1/1 or 1/3 octave and **Compute bands** if you need spectral decay diagnosis.
- Export CSV or IR WAV for reports.

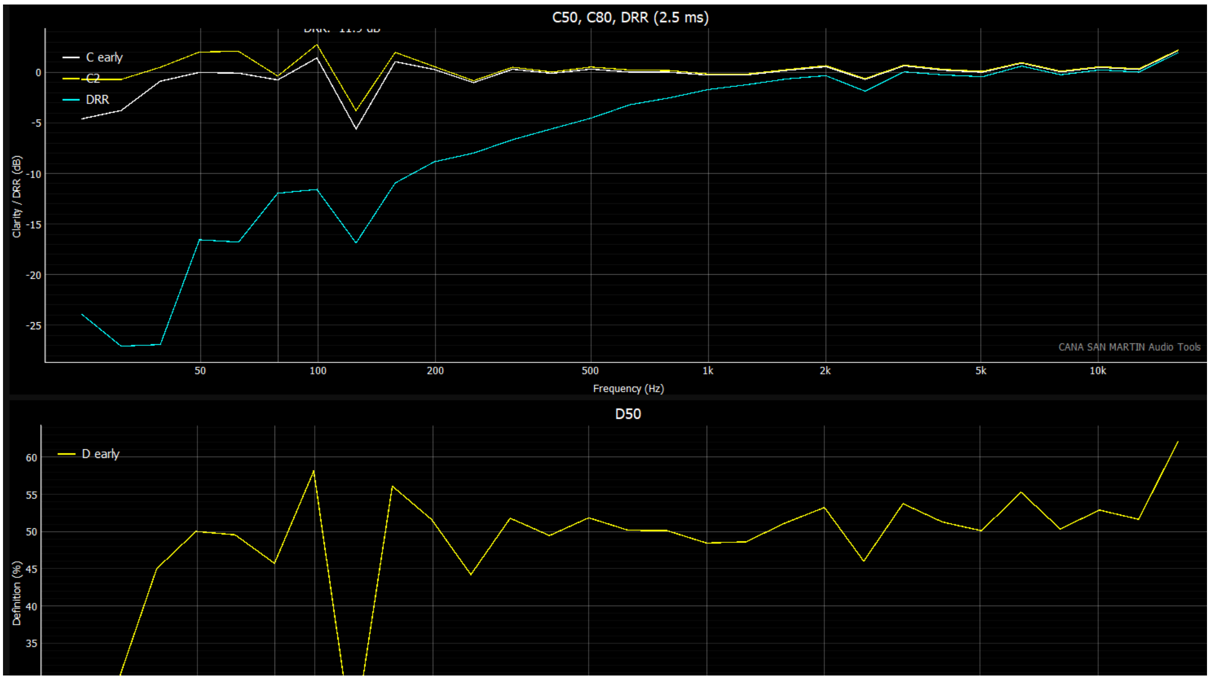
7. Pestaña Acoustic Graphs

The **Acoustic graphs tab** is designed to visualize, **as a function of frequency**, the classic acoustic room parameters (decay times, clarity, definition, etc.) derived from the **impulse response (IR)** reconstructed from the **transfer function (TF)**. In **SMartin**, the baseflow is:

- **a TF** (magnitude + phase) is obtained ,
- **An IR** is reconstructed by **IFFT**, with the option to align the direct sound **peak to** *t = 0* ,

- The Schroeder -type **energy decay curve (EDC)** is calculated .
- **Broadband** parameters are extracted , and (optionally) **by bands** (1/1 or 1/3 octave), which are precisely what are then plotted in **Acoustic graphs** .
  This is an explicit part of the design of the room parameters module in the project's codebase.





7.1 Common Foundation: IR, Energy and DC EDC (Schroeder)

7.1.1 Impulse Response (IR)

The **IR** is the "temporal footprint" of the *speaker + room + microphone system* :

- **Initial peak** : direct sound (and/or first arrival).
- **Early reflections** : first relevant reflections (walls, console, floor/ceiling).
- **Reverberating tail** : diffuse field and global decay.

At **SMartin** , room calculations are based on instantaneous **energy :**

$$E(t) = h^2(t)$$

The module also considers **peak alignment** to place the live signal at $t = 0$ (which is crucial for metrics based on "early/late" windows such as C50/D50/DRR).

7.1.2 Energy decay curve (EDC / Schroeder)

The **EDC** is obtained by **inverse integration** of the energy:

$$EDC(t) = \int_t^{\infty} E(\tau)\, d\tau$$

In discrete terms, **SMartin** implements this as a cumulative sum "from the end" (equivalent to Schroeder's method), and expresses it in dB relative to the start.

What it means physically:
EDC describes how much **residual acoustic energy** remains in the room at any given time. If the room is approximately diffuse over a range, EDC tends to be **nearly linear** when plotted as dB vs. time.

7.1.3 Noise, dynamic range and decay reliability

RT metrics (T20/T30, etc.) depend heavily on the EDC having sufficient "drop-off" **above the noise floor** . SMartin calculates:

- **Noise estimation** (e.g., from the final tail of the EDC).
- **Minimum EDC** and **dynamic range** of decay (approx. how many dB the EDC drops before hitting the noise).

And it issues warnings when the EDC does not reach the necessary ranges (for example, to estimate T30 it is necessary to reach close to −35 dB).

Why this matters: If noise "cuts" the tail, linear fitting can skew the result (typically **overestimating** RT), because the EDC stops falling and "flattens."

7.2 Acoustic parameters that SMartin evaluates in this family (and that feed into Acoustic graphs)

SMartin explicitly defines the main set of parameters "Room acoustic parameters" and also the option of calculating by bands (octaves/thirds), which is what **Acoustic graphs** usually represents as curves vs frequency.

Next, the theory, calculation, and use of each parameter.

7.2.1 Decay times: EDT and RT60 (T10/T15/T20/T30)

Idea general

The EDC is taken in dB and a straight line is fitted:

$$EDC_{dB}(t) \approx a\,t + b$$

where is the slope in dB/s.
If the decay were perfectly exponential, this line would be very stable.

Then it is extrapolated to a decay of 60 dB:

$$RT60 = \frac{-60}{a}$$

In **SMartin** , the conversion "slope → RT60" is done exactly like this (−60/slope).

7.2.2 EDT (Early Decay Time)

It is an estimator of the "perceived reverberation time" dominated by the first part of the decay (highly correlated with "live/dry" perception).

How it's calculated: Linear fit over the **first 10 dB** of drop (0 to −10 dB) and extrapolation to 60 dB. **SMartin** uses a fit over that range:

- Typical range: dB.
- Final result: .

What is it for:

- Sensitive to early reflections on how the room "starts".
- Very useful in small rooms where the field is not diffuse and T30 may not be representative.

7.2.3 T10 / T15 / T20 / T30 (RT60 family by segments)

They are RT60 estimators based on different sections of the EDC (with different dynamic range requirements).

How they are calculated in SMartin: Linear fits in standard ranges:

- **T10:** −5 a −15 dB
- **T15:** −5 a −20 dB
- **T20:** −5 a −25 dB
- **T30:** −5 to −35 dB
  and then extrapolation to 60 dB in all cases.

What are they for:

- **T30** is usually the most "classic" choice for RT60 when there is sufficient dynamic range.
- **T20** is a common compromise when the measurement does not reach −35 dB.
- **T10/T15** are useful when the queue is limited by noise or when the room is very dry.

Practical reliability criterion (very important):
If the EDC does not reach the end of the segment, the value is unreliable. SMartin detects and marks this (e.g., if the minimum EDC does not reach −35 dB, T30 is "not reliable").

7.2.4 Fit quality

SMartin calculates a quality of fit metric (R^2) on the used segment, based on the squared error and variance of the segment:

$$R^2 = 1 - \frac{\Sigma(y-\hat{y})^2}{\Sigma(y-\bar{y})^2}$$

As shown in the implementation of the adjustment.

How to interpret it:

- **R^2 close to 1** : approximately exponential decay in that section → more reliable RT.
- **Low R^2** : non-linear decay, noise, dominant reflections, or highly modal room → interpret with caution.

7.2.5 Clarity: C50 and C80

Clarity **compares** early vs late energy:

$$C_x = 10log_{10}\left(\frac{\int_0^x E(t)\,dt}{\int_x^\infty E(t)\,dt}\right)$$

- **C50** :$x = 50$ ms (speech intelligibility oriented).
- **C80** : $x = 80$ ms (oriented towards musical clarity).

How SMartin calculates it:

- Sum energy up to a time index **early_ms** and compare it with the later (late) energy.
- The same applies to **clarity2_ms** .
  The code explicitly shows the early/late sum and the calculation with log10.

Furthermore, the label is constructed as `C{early_ms}` and `C{clarity2_ms}` , confirming that in **SMartin** these are configurable parameters (by default they are usually 50 and 80 ms).

What is it for:

- **C50** : strong predictor of intelligibility (more early energy relative to the tail → better speech clarity).
- **C80** : describes musical definition (attacks, note separation, articulation).

Reading in Acoustic graphs:
Seeing C50/C80 by band helps to understand if the room "muddies" by frequency (for example, low clarity in bass due to modal tail, or drop in mids due to excess reflections).

7.2.6 Definition: D50

The **definition** is the fraction of early energy within a window (classically 50 ms):

$$D_{50} = 100 \cdot \frac{\int_{0}^{50ms} E(t)\, dt}{\int_{0}^{\infty} E(t)\, dt}$$

How SMartin calculates it:
It is calculated as a percentage of early energy over total energy (100×early/total).

What is it for:

- Widely used in **speech assessment** and rooms where "how much useful energy arrives quickly" matters.
- In **Acoustic graphs** , D50 by frequency is extremely illustrative for diagnosing loss of definition in specific bands.

7.2.7 Central Time: Ts (Center Time)

Central time is the “temporal center of mass” of energy:

$$T_s = \frac{\int_{0}^{\infty} t\, E(t)\, dt}{\int_{0}^{\infty} E(t)\, dt}$$

How SMartin calculates it:
It is implemented exactly as a time-weighted sum over total energy.

Interpretation:

- **Low Ts** → concentrated energy early → more “direct” room / more clarity.
- **High Ts** → energy shifted towards the tail → more perceived reverberation and/or dominant late reflections.

What is it for:

- Complements C50/C80: where clarity increases, Ts usually decreases.
- In **Acoustic graphs** , Ts per band allows you to identify bands where the system "retains" energy (typical in bass).

7.2.8 DRR: Direct-to-Reverberant Ratio

The **DRR** (direct/reverberant ratio) is:

$$DRR = 10 log_{10}\left(\frac{E_{directo}}{E_{reverberante}}\right)$$

The challenge lies in defining what is considered "direct". In SMartin, a configurable **drr_ms window is used** : energy from 0 up to that limit is considered "direct"; anything beyond that is considered "reverberant".

How SMartin calculates it:
He sums up to a `drr_ms index` and compares it to the remaining energy, in dB.

What is it for:

- Strong indicator of **perceived distance** and “dry vs. ambient”.
- Very useful for comparing positions: moving the microphone 20–40 cm can change DRR and explain subjective changes.

Practical care:

- In small rooms, very early reflections can "contaminate" the live performance if the window is too long.
- In large venues, a very short window can undervalue the live performance.

7.2.9 ITDG and early reflections (ETC / reflection table)

Although not strictly a “band chart”, SMartin also derives and reports **early reflection information** , including:

- **Table of reflections** with relative times and levels,
- **ITDG (Initial Time Delay Gap)** as the time between the live signal and the first meaningful reflection.

In the implementation it is observed that, after detecting reflection peaks, the ITDG is taken as the positive minimum after the direct (dt_ms) and is reported in ms.

What does ITDG mean?

- A **larger gap** tends to favor a sense of "space" without cluttering the image, because the brain separates directly from reflections (Haas effect/precedence).
- A **very short gap** can degrade localization and stereo image (reflections too close to the direct source).

What is it useful for in practice:

- Fine diagnosis of **first reflections** (console, sides, ceiling).
- Decide on location and type of treatment (absorption/diffusion) to control *timing* and level of those reflections.

7.3 How to read Acoustic graphs correctly (practical criteria)

1. **Don't look for a single "magic" number.**
   What's valuable is the **frequency profile** : how EDT/RT/C50/C80/D50/Ts/DRR vary in each band.
2. **Relate parameters to each other:**

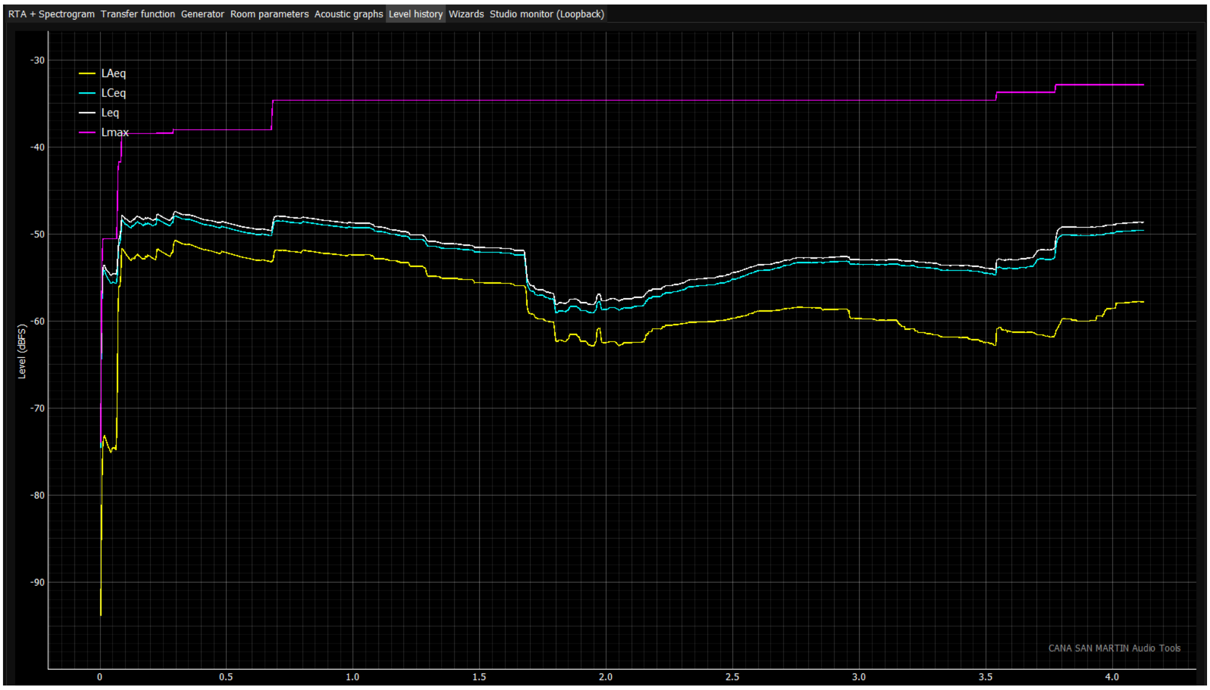
   a. High RT in bass + high Ts in bass usually indicates **modal ringing** .
   b. Low C80 + low D50 generally coincides with a perception of "muddy".
   c. Low DRR where C80 also falls is usually a symptom of a lot of late energy and/or dominant reflections.

3. **Always check reliability (noise/dynamic range/R^2).**
   If SMartin indicates that T20/T30 are unreliable due to insufficient EDC drop, use EDT/T10/T15 and repeat the measurement, improving the SNR.

7.4 Measurement recommendations to make Acoustic graphs "scientific" and repeatable

- **Real SNR:** raise the excitation level without clipping, and measure silently.
- **Live alignment:** if the live feed is not properly identified, C/D/DRR are distorted (that's why the peak-align option is important).
- **Geometric consistency:** document microphone and speaker position; small variations generate real differences.
- **Multi-position (average):** If you're looking to represent "the room" and not a point, averaging positions is more robust (especially in LF).
- **Bands:** for fine diagnosis use 1/3 oct; for global reading use octaves.

8. Level History Tab



8.1. General purpose

The **Level History tab** is designed for **recording acoustic levels over time**, allowing:

- Continuous monitoring.
- Regulatory compliance.
- Event analysis.
- Generation of technical evidence.

It is a **fundamental tool in legal, labor and entertainment contexts**.

8.2 . What does the main graph show?

8.2.1 Typical plotted parameters

- **LAeq**
- **LCeq**
- **Leq (flat)**
- **Lmax**
- (according to configuration)

Each parameter is represented as an **independent curve** over time.

8.2.2 Axes

X-axis: Absolute or relative time (hh:mm:ss).

Y-axis: Level in dBFS or dB SPL (depending on calibration).

8.3 . Temporal Integration

8.3.1 Theoretical basis

The **equivalent level** is defined as:

$$L_{eq} = 10log_{10}\left(\frac{1}{T}\int_{0}^{T}\frac{p^{2}(t)}{p_{0}^{2}}dt\right)$$

Where:

- T It's the integration time.
- $p(t)$ instantaneous sound pressure.

8.3.2 Integration Control

The user can select:

- 1 s
- 10 s
- 1 min
- Custom Windows

Impact

- Short integrations → greater sensitivity to transient events.
- Long integrations → better exposure assessment.

8.4 . Logging (logging to file)

8.4.1 Operation

During a recording session:

- Levels are **automatically recorded** to disk.
- The writing interval is configurable (e.g., 0.5 Hz).

8.4.2 Generated Files

- CSV or other structured format.
- Timestamp + values per parameter.
- Compatible with audits and reports.

8.5 . Scoreboards and events

8.5.1 Manual markers

They allow:

- Highlight relevant events (show start, test, peak).
- Relate levels to concrete actions.

8.5.2 Visualization

The markers appear as:

- Vertical lines in the graph.
- Associated textual annotations.

8.6 . Regulatory compliance

8.6.1 What it evaluates

The system compares:

- LAeq
- Lmax
- Duration

Against configured thresholds (according to local or international regulations).

8.6.2 Visual indicators

- OK / WARNING / ALARM states.
- Event latch.
- Persistent alarms.

8.7 . Typical field use

- **Live events** : audience exposure control.

- **Work environments** : hygiene and safety.
- **Audits** : generation of traceable evidence.
- **Studies** : dynamic content analysis.

8.8 . Practical recommendations

- Always calibrate before recording.
- Define integration in accordance with regulations.
- Activate logging before the event.
- Use markers to contextualize data.

9. Wizards Tab

The **Wizards tab** focuses on task-oriented **workflows** designed to **reduce human error**, **accelerate repetitive tasks**, and **standardize measurement procedures** in real-world audio system tuning and verification situations.

Unlike the "free" tabs (RTA, TF, etc.), the Wizards:

- They impose a **logical order of steps**,
- They apply **automatic validity criteria**,
- and generate **objective evidence** of the result.

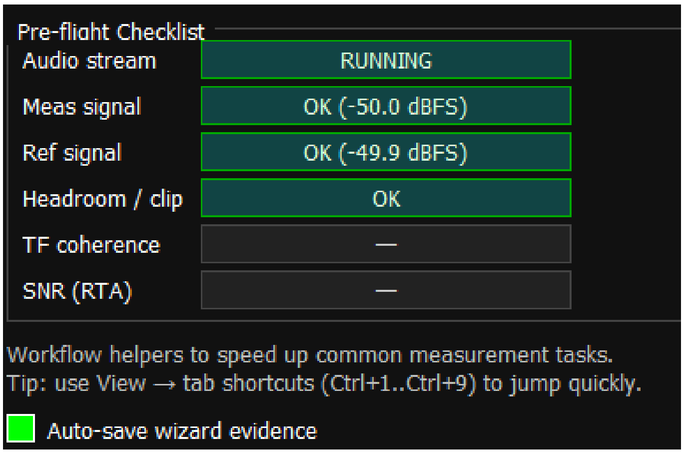
At **SMartin**, Wizards are designed as the **bridge between pure measurement and technical decision-making**.

The available Wizards include:

- Polarity Check
- Sub alignment (sub delay)
- Multi-position RTA average
- Fill alignment (delay + level + polarity)
- Multi-position TF average
- Quick capture (RTA + TF + quick report)

This block transforms **SMartin** not only into a measurement tool, but into a **professional work system**, aligned with real sound engineering practices.

9.1. Pre-flight Checklist



The **Pre-flight Checklist** is a **continuous verification system for the measurement chain**, visible in the right-hand column. It is not a wizard itself, but a **quality assurance (QA) module** that reports, in real time, whether current conditions allow for **reliable measurements**.

It functions as a "technical traffic light" prior to:

- Run Wizards,
- capture traces,
- export reports,
- or make adjustment decisions.

What it evaluates: The checklist simultaneously analyzes several critical states:

1. **Audio stream**
    a. Verify that the audio engine is active and receiving valid blocks.
    b. Detects stopped streams, backend errors, or blockages.
2. **Meas signal (measurement level)**
    a. Evaluate whether the measurement signal has a sufficient level.
    b. Detects levels that are too low (dominant noise) or invalid.
3. **Ref signal**
    a. Confirms presence and stability of the reference (TF).
    b. Essential for coherence, polarity, and delay.
4. **Headroom / Clip**
    a. Calculate available dynamic margin.
    b. Mark **FAIL** if there is clipping, **WARN** if the headroom is insufficient.
5. **TF coherence**
    a. Use average consistency metrics.
    b. Indicates whether a transfer function is physically interpretable.
6. **SNR (RTA / measurement)**
    a. Estimated signal-to-noise ratio.
    b. Avoid interpreting spectra dominated by ambient noise.

Each item is displayed as:

- **OK** (green)
- **WARN** (yellow)
- **FAIL** (red)

and it updates automatically several times per second.

Practical use

The Pre-flight Checklist **does not block** actions, but:

- **warns** when a measurement may be misleading,

- **educate** the operator about what is failing,
- and serves as a **technical compliance check** before documenting results.

9.2 Auto Save Wizard Evidence

The Auto Save Wizard Evidence option (on the Wizards tab) enables **automatic saving of technical evidence** each time a Wizard is run.

When active, **SMartin** :

- saves the results **without user intervention** ,
- avoid forgetting critical captures,
- and generates complete traceability of the process.

According to the Wizard, evidence may include:

- **JSON** files with numerical results and metadata,
- **PNG images** of graphics (RTA, TF, goniometer, etc.),
- timestamps and tags of the executed Wizard.

The files are stored in the **project's generated file structure** , organized by date and measurement type, and can be opened directly from the **Open evidence folder button** .

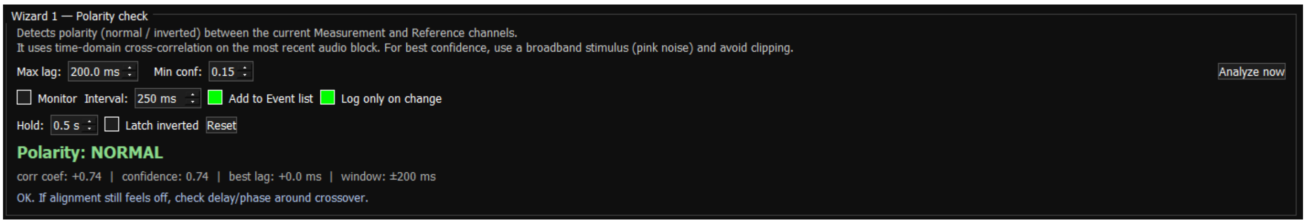
It is especially useful in:

- professional system adjustments,
- auditable work,
- regulatory compliance,
- post-show or post-installation documentation.

9.3 Wizard 1 – Polarity Check

The **Polarity Check** determines if the **acoustic polarity** between the measurement signal and the reference signal is:

- **NORMAL** (in phase),
- **INVERTED** (180°),
- or **UNKNOWN** (insufficient trust)



Correct polarity is critical for:

- coherent sum between paths,
- sub alignment,
- stereo image consistency,
- and low-frequency response.

9.3.1 Theoretical basis

The Wizard is based on **temporal cross-correlation** between:

- the measurement signal block,
- and the reference signal block.

The simplified procedure is:

1. DC components are removed.
2. The full cross-correlation is calculated.
3. The peak correlation is sought within a **maximum delay window** (in ms).
4. The value is normalized:

$$\rho = \frac{max(|R_{xy}|)}{||x|| \, ||y||}$$

5. If it exceeds the confidence threshold:
    a. positive sign → **NORMAL**
    b. negative sign → **INVERTED**

Otherwise, the result is **UNKNOWN** .

9.3.2 Wizard Controls

The operator can adjust:

- **Max lag (ms)**
  Maximum peak search window (useful in systems with physical delay).
- **Confidence threshold:**
  Minimum acceptable correlation value for declaring a result.
- **Hold time (s)**
  Time during which the result must remain stable before being displayed.
- **Latch inverted**
  Allows you to "latch" the INVERTED state until it is manually reset.
- **Monitor polarity.**
  Runs the analysis continuously (not just in one shot).

9.3.3 Result and interpretation

The Wizard displays:

- textual state (NORMAL / INVERTED / UNKNOWN),
- correlation coefficient,
- estimated delay,
- and a **suggested action** (e.g., reverse electrical polarity).

This transforms an abstract concept into a **clear and verifiable technical decision** .

9.4 Wizard 2 — Sub Alignment

What does it do (module function)

Wizard 2 calculates an **estimate of the relative delay between two Transfer Function (TF)** measurements (traces) around a user-defined **crossover frequency . In the flow specified by SMartin** , it is recommended **to capture the "Main" as trace A** and the **Sub as trace B** , and then run the wizard. Additionally, you can apply **coherence weighting** to stabilize the calculation when TF quality varies with frequency.

In operational terms, the wizard aims to answer:

- "How many milliseconds should I delay (or advance, if possible) the SUB with respect to the MAIN so that they sum correctly at the crossover?"
- "Is the relative polarity correct, or should it be reversed to maximize the sum?"

Theory (what it is measuring and why)

Main/Sub alignment is based on the fact that, in the crossover region, the acoustic result is the **vector sum** of both sources. If, in that region, the **relative phase** (or equivalently, the **relative time**) is not aligned, the following occurs:

- **Partial or total cancellation** in the crossover (notch/“hole”).
- **Irregular response** (phase combs) and loss of impact/definition.
- Worse translation to the rest of the positions.

Phase shift (PSS), conceptually, is a complex relationship between a reference signal and the measured signal. A **delay** manifests as a **linear phase slope** (phase proportional to frequency). Therefore, estimating relative delay by comparing two PSSs is a robust approach.

“Consistency weighting” is important because the estimated delay can be degraded if there is:

- high ambient noise,
- poor signal-to-noise ratio,
- dominant reflections,
- saturation or insufficient headroom,

- reference problems (clocking, routing, etc.).

Under these conditions, consistency helps to give more weight to reliable bins.

9.4.2 Complete procedure (in an actual setting)

Below is a complete and operational procedure (standard in PA) for using Wizard 2 correctly:

1) System preparation

- **Ensure routing and muting:** have independent control (mute) of MAIN and SUB in the DSP/console.
- **Check electrical polarities:** that the wiring is correct, and that there is no accidental reversal in one of the paths.
- **Select stimulus:** noise (ideally **band-limited** around the crossover) or a suitable generator to increase coherence in the critical band.
- **Adjust levels:** the SPL should be sufficient to dominate ambient noise without saturating inputs/outputs.

2) Selection of the measurement point (mic placement)

1. For **sub-alignment** , the measurement point must be where:
    a. MAIN and SUB are **simultaneously present** at a comparable level around the crossover,
    b. Don't be too close to a sub (avoid extreme near-field),
    c. be representative of the main area (a common practice: position close to FOH or a point representative of the system's "average").

Rule of thumb: if you measure too close to a source, the result becomes local and does not generalize.

3) TF capture (trace)

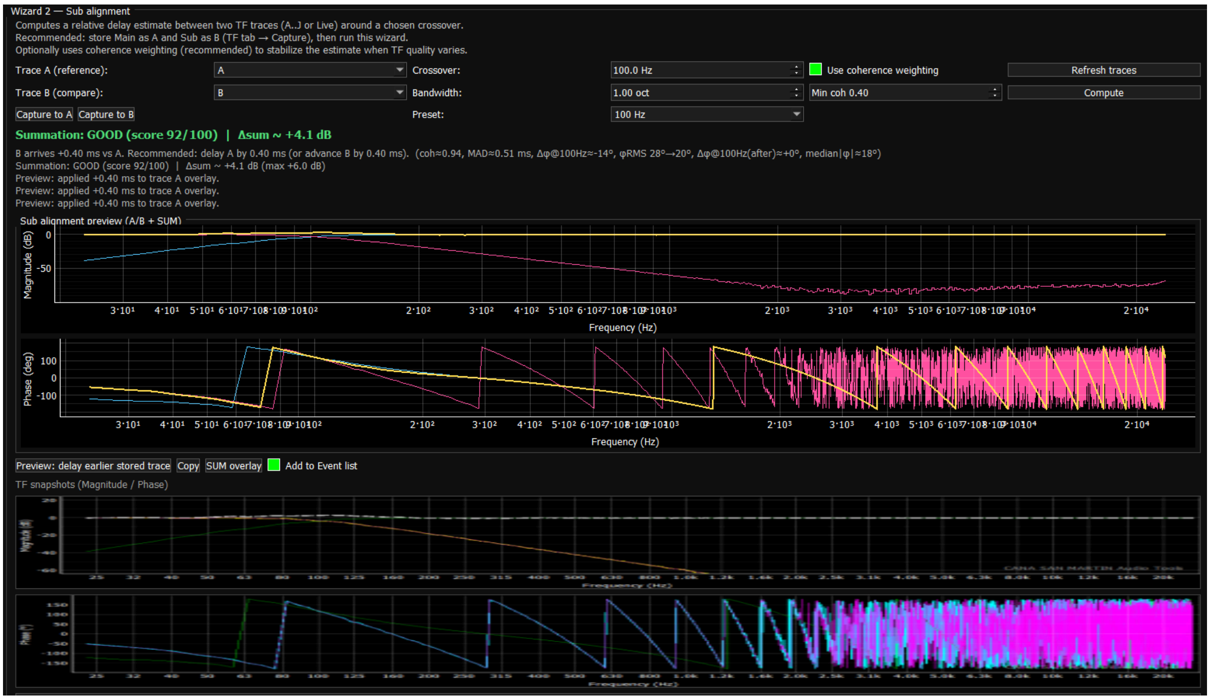
- **Mute SUB** , leave MAIN alone.
- In the TF tab, stabilize and **capture** the TF of the MAIN as **Trace** A.
- **Mute MAIN** , leave SUB alone.
- Capture the TF of the SUB as **Trace** B.

*(**SMartin** even includes "Capture to A / Capture to B" helpers within the wizard, so as not to leave the workflow.)*

4) Wizard configuration and calculation

1. Abrir **Wizard 2 — Sub alignment**.
2. Seleccionar **Trace A (reference)** = MAIN y **Trace B (compare)** = SUB.
3. Define **Crossover** (nominal frequency of the system or DSP preset).
4. Activate **coherence weighting** if available/recommended (especially in noisy venues or with strong reflections).

5. Run the calculation.



5) Correction application

1. **Apply the recommended delay** to the appropriate path (typically in SUB, although it depends on the system design).
2. If the wizard suggests polarity reversal, **try reversing** on SUB (or MAIN) and compare.

6) Verification (mandatory step)

1. **Unmute MAIN + SUB**.
2. Measure TF of the **SUM** .
3. Confirm:

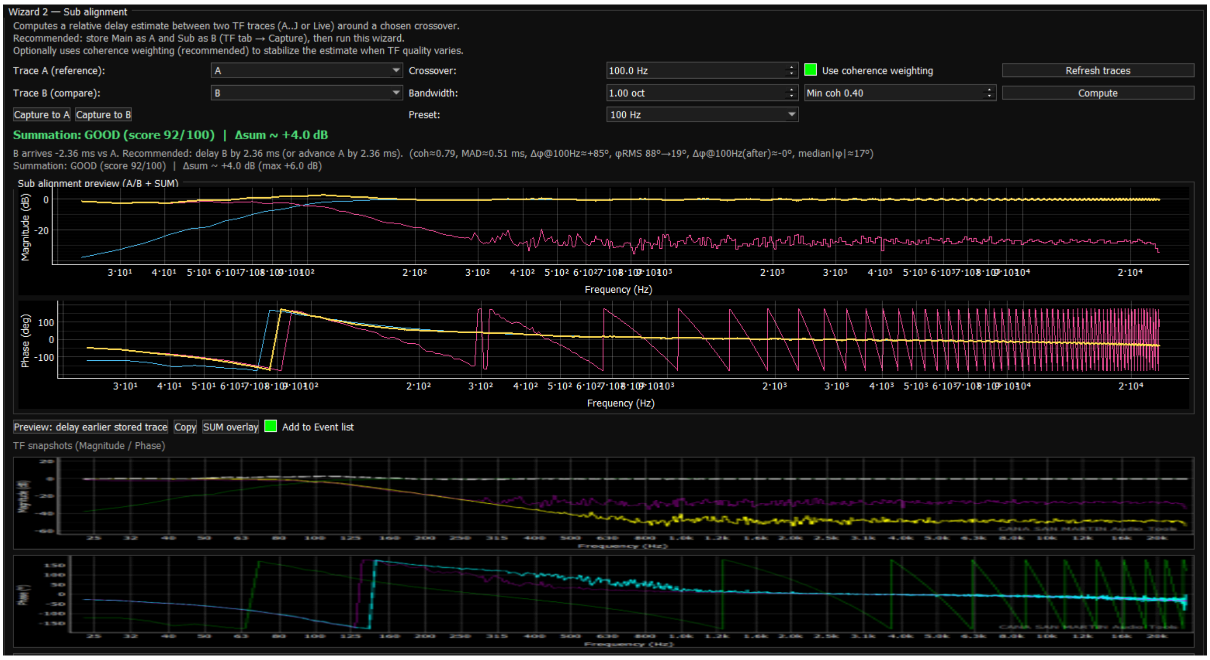
   a. **Magnitude** without notch in the crossover.
   b. **Phase** aligned as closely as possible around the crossover band.
   c. Reasonable consistency in the critical zone.

7) Spatial robustness (not relying on a single point)

1. Repeat the verification in 2–4 relevant positions:

   a. near the center,
   b. a couple of side points,
   c. some point closer/farther if the system warrants it.

2. Adjust towards a **compromise solution** (perfect alignment at one point may worsen another if the geometry is complex).



9.5 Wizard 3 — Multi-position RTA average

What does it do (module function)

This wizard allows you to capture **N RTA spectra** in different positions and calculate a **power-average** , storing the result as an RTA trace (A..J) for comparison.

In practical terms, the answer is:

- "What is the average spectrum for this area (multiple seats/positions), instead of a single local measurement?"
- "How do I quickly compare the 'room average' against a target or another system?"

Theory (why averaging in power)

A common error in system tuning is "equalizing for one point". Multi-position averaging seeks to approximate spatially representative behavior .

To average spectra, doing it "by eye" in dB is incorrect. The reason:

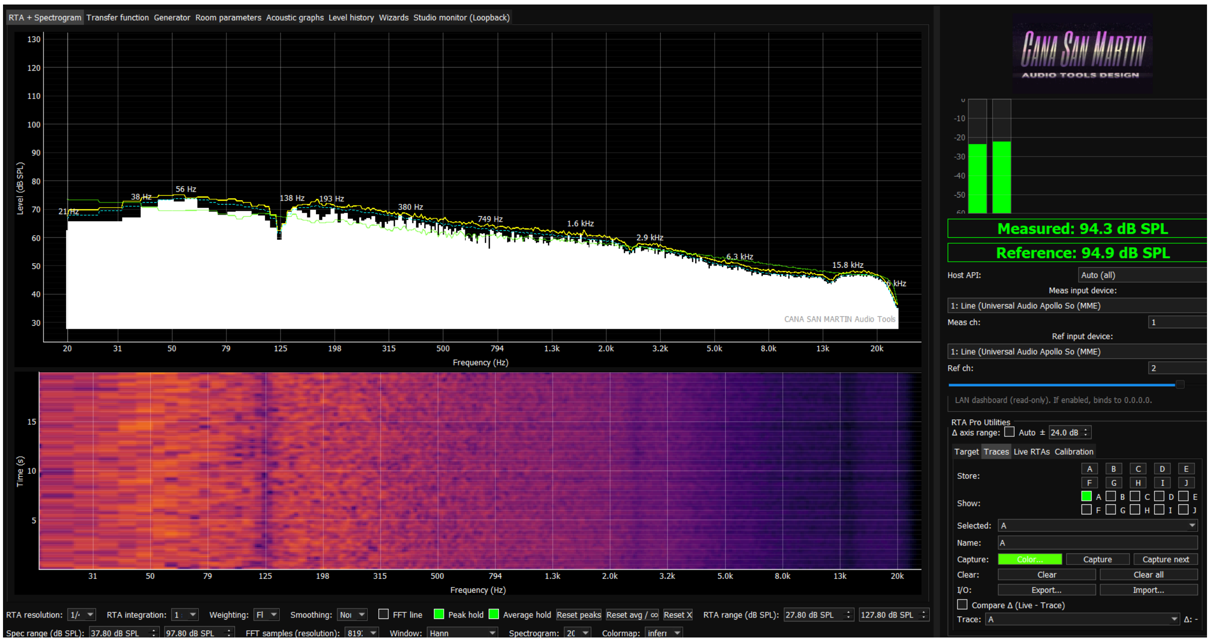
- dB is logarithmic.
- The correct energy average is done in the linear domain (**power/amplitude**), and then converted back to dB.

The wizard explicitly states "power-average", which is consistent with this need.

9.5.2 Complete procedure (in an actual setting)

1. Define the **target area** : FOH, central seating, VIP section, etc.
2. Select 5 to 12 positions (depending on available time):
   a. cover center and sides,
   b. include 1–2 critical positions (e.g., “front rows” and “transition zone”).
3. In each position:
   a. stabilize reading (adequate integration time),
   b. capture spectrum.
4. Run the wizard to generate the average.
5. Save the average as a trace (A..J) and use it for:
   a. global EQ decisions,
   b. comparison against targets,
   c. view trends (excess of LF, lack of HF, etc.) without local bias.

Important: RTA averaging does not replace TF when it comes to time alignment; it is complementary for spectral balancing.



9.6 Wizard 4 — Fill alignment (frontfills / outfills)

This wizard uses **TF (magnitude and phase)** around a chosen crossover to **recommend delay and polarity** between a main system and a fill. The included operating guide states:

- Measure in a position where both can be heard (overlap zone),
- apply the suggested delay to the fill channel,

- verify with the SUM (overlay) measurement.

Fills (frontfills/outfills, etc.) usually cover areas where the main system:

- It doesn't reach a sufficient level,
- loses intelligibility due to geometry,
- It needs local reinforcement.

But if fill and main overlap without alignment:

- **comb filtering**,
- intelligibility degrades (especially in the 1–4 kHz range),
- The stereo image/perceived location becomes unstable.

Aligning fills involves adjusting the **relative time** (delay) and, where appropriate, the **polarity** , to maximize coherent summation in the transition zone.

9.6.2 Procedure

1) Define "where" to align (decision point)

For fills, the critical point is not necessarily FOH. The usual practice is to choose:

- the **limit** of the fill's coverage area (where the main starts to "command"), or
- a representative **overlap zone** .

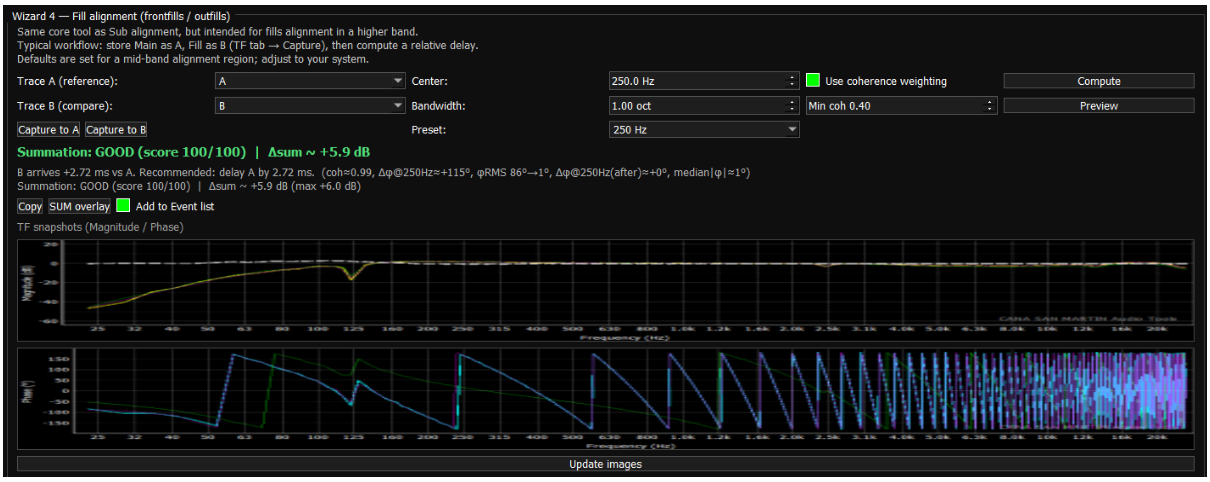
This produces a smoother transition in the actual area of use.

2) Preparation

1. Ensure that the fill has:
    a. reasonable nominal level,
    b. protection filters and/or correct crossover,
    c. independent routing and muting.
2. Select stimulus:
    a. band-limited noise in the band of interest (very useful),
    b. or an equivalent generator that allows good coherence.

3) Capture TF traces

1. **Solo MAIN** → capturar TF (Trace A).
2. **Solo FILL** → capturar TF (Trace B).
3. Run wizard with the crossover or transition frequency/band.



4) Apply delay and polarity (and respect precedence)

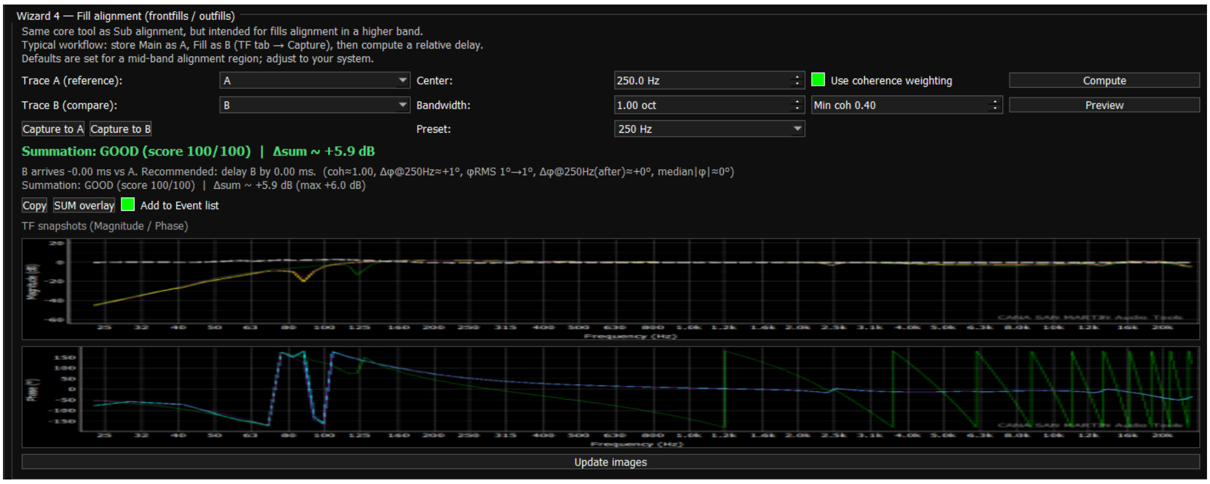
- Apply delay to the **fill** to bring its arrival time closer to that of the main at the chosen point.
- In addition to maximizing feedback, there's a psychoacoustic criterion for fills: **preventing the fill from taking over the** desired location.
  In many scenarios, the goal is for the main sound to be perceived as dominant, which means not advancing the fill (and typically delaying it enough for coherence without causing an echo).

5) Verification (SUM)

1. With MAIN+FILL active:
   a. check magnitude in overlap band,
   b. review relative phase if applicable,
   c. Listen to real material (voice) to validate that the "measured" solution matches perception.

6) Spatial validation

- Move 2–5 positions along the edge of the overlap (left/right and slightly forward/backward).
- Adjust towards a compromise if a very marked "comb" appears in a sector.



9.6.3 Recommended workflow in a setting (logical order)

A robust field procedure, using these wizards as tools, typically follows this order:

1. **Pre-flight** : routing, references, headroom, basic consistency.
2. **Polarity check** (Wizard 1, if applicable) to ensure the system starts from a correct base.
3. **Sub Alignment (Wizard 2)** : align SUB with MAIN at the intersection.
4. **Fill alignment (Wizard 4 / "Fill Alignments")** : align frontfills/outfills against the main system, using the correct overlap point.
5. **Multi-position RTA average (Wizard 3)** : obtain a representative spectrum and make global EQ and/or target matching decisions with less bias.
6. Final verification with real material + walk test.

9.7 Wizard 5: Multi-position TF magnitude average

This wizard captures multiple **Transfer Function** (TF) measurements at various positions and calculates the **spatial average of the magnitude** (not the phase, unless you implement a phase-specific method).

At the computational level, the magnitude average is performed as follows:

- **A reference grid** is defined (frequencies of the first capture).
- Each capture is ordered by frequency and interpolated to the reference grid **on the logarithmic axis** (`log10(f)`), which is correct for audio data (perceptual spacing/thirds/octaves).
- Gating is applied **by consistency : only bins whose consistency exceeds a** `min_coh` threshold are averaged .
- The magnitude in dB is converted to linear (amplitude) `10^(dB/20)` , averaged, and converted back to dB with `20·log10` .

In addition, it can maintain/accompany an average of coherence as a diagnostic reference.

Finally, the result is stored as an external TF trace (e.g., “TF AvgMag”) and the UI reports it as *Stored TF AvgMag -> slot* .

The TF (with reference) tends to separate better:

- system response,
- of the signal content.

Therefore, the multi-position TF average is usually **more "adjustable"** than an RTA (which is absolutely valid, but more dependent on the program and the direct/reverberant field).

Gating for consistency prevents the average from being contaminated by:

- low coherence due to noise, distortion, wind, audience, etc.
- bands where the signal-to-noise ratio or the linearity of the system do not allow for a reliable TF.

9.7.2 Procedure

1. **Assemble TF correctly** (ref and meas properly routed, synchronized, levels).
2. Measure in the first position and verify:
    a. reasonable consistency within the working range,
    b. that the excitation spectrum covers the bandwidth,
    c. that there is no clipping or lack of headroom.
3. Define Minimum Coherence:
    a. If you are in a difficult field/environment: a lower threshold may allow you to see more range, but with a greater risk of contamination;
    b. If you are in controlled conditions: a higher threshold produces a more reliable average.
4. Capture the positions (same spatial logic as above).
5. Use the magnitude average for:
    a. EQ tonal global,
    b. see trends by band (e.g., consistent 200–400 Hz overshoot),
    c. Evaluate coverage consistency if you compare with traces by position.

9.8 Wizard 6 - Quick capture (RTA + TF)

It's a "fast" stream that captures **RTA and TF together** per position and then calculates both averages. When capturing a point:

- If there is no RTA data yet, it displays a message like "No RTA data yet..." (you have to start audio and wait for spectrum).
- If there is no TF data yet, it displays “No TF data yet…” (TF measurement must be started).

Each capture adds:

- snapshot RTA (frequency + dB)
- snapshot TF (frequency + magnitude + phase + coherence)

Then, when calculating averages, apply:

- spatial average of the RTA in **linear power** and return in dB.
- spatial average of the TF by **log-f interpolation** , **coherence threshold** and average in linear amplitude.

And it has an action to **save both results** (RTA AvgPow and TF AvgMag) in their corresponding “slots”/traces, using:

- `store_external_trace(... kind="avgpow")` para RTA
- `store_external_mag_trace(...)` for TF

Quick capture doesn't invent a new theory: it simply packages it into a single stream:

- **spatial energy average** (RTA),
- **spatial average magnitude of the system** (TF, with coherence).

which is the correct way to average spectra:

1. Convert dB → linear:

$$P = 10^{\frac{dB}{10}}$$

2. Linear average:

$$P^{-} = \frac{1}{N}\sum_{i=1}^{N} P_i$$

3. Back to dB:

$$dB^{-} = 10log_{10}(P^{-})$$

Why is this correct? Averaging "in direct dB" is mathematically incorrect for power. Power averaging avoids bias and better represents the "acoustic average" per band. This is useful when you want a "quick summary" of coverage/tonality for an area without performing two separate procedures.

9.8.2 Recommended procedure

1. Use it when you already have it :
    a. RTA audio/routing operation,
    b. TF running with decent consistency.
2. Define **positions** before starting and maintain a pattern of movement (e.g., meandering through rows, ideally 6 to 20 positions).
3. Do not capture while walking: stop for 1–3 s (depending on integration) and make a stable capture.
4. Finally, save both averages as traces and compare them against:
    a. target (if applicable),
    b. critical point measurements (FOH, mix position, etc.).

9.8.3 TF min coh

- Range: **0.00 to 1.00**
- Typical default: **0.40**
- **Define a minimum coherence** threshold for the TF average.

When averaging the **TF magnitude** , the wizard **ignores** (does not average) frequencies where the coherence is below that value. This prevents the average from being "contaminated" by bins where the measurement is unreliable (low SNR, poor reference, distortion, etc.).

Recommended use:

- 0.30–0.50: good compromise for real field.
- 0.60+: only if you have a very stable measurement (very good SNR).
- 0.00: if you absolutely want to average everything (not recommended live).

Controls:
9.8.4 Clear

- Delete accumulated captures (reset the position set).
- Useful if you made a mistake in the location or changed an important parameter and want to start over.

9.8.5 Compute averages

- Force the calculation of the average at that moment.
- Requires a minimum of 2 captured **positions .**

9.8.6 Store RTA / Store TF + Store both

- `Store RTA` : Choose the slot (A–J) where the **average** RTA trace will be stored.
- `Store TF` : Choose the slot (A–J) where the **average** TF trace will be stored.
- `Store both` : saves both results at the same time, with a name like **QuickAvg(n)** .

9.8.7 Add to Event list

1. If enabled, when computing/saving, it adds an **event/marker** of the type:
   a. "Wizard: Quick averages computed (n pos)"
2. That marker then appears in the **Event List** and in **Level History** as a time reference (useful for reports and traceability).

9.8.8 Export PDF…

- Triggers the system's PDF export.
- Practical recommendation: Use it **after** "Store both", so the report includes the saved and visible averages for reference.

9.8.9 Common Errors and Troubleshooting

1. **"No RTA data yet"**
   a. Audio did not start or the RTA has not yet received enough blocks.
2. **"Could not capture TF snapshot"**
   a. TF is not producing valid arrays at this time (unstable measurement, motor stopped, no signal/ref).
3. **"Strange" or inconsistent average**
   a. You were capturing with configuration changes between positions (FFT size, smoothing, etc.)
   b. You were moving the mic while you were recording.
   c. Very low coherence: temporarily increase SNR or decrease TF min coh .

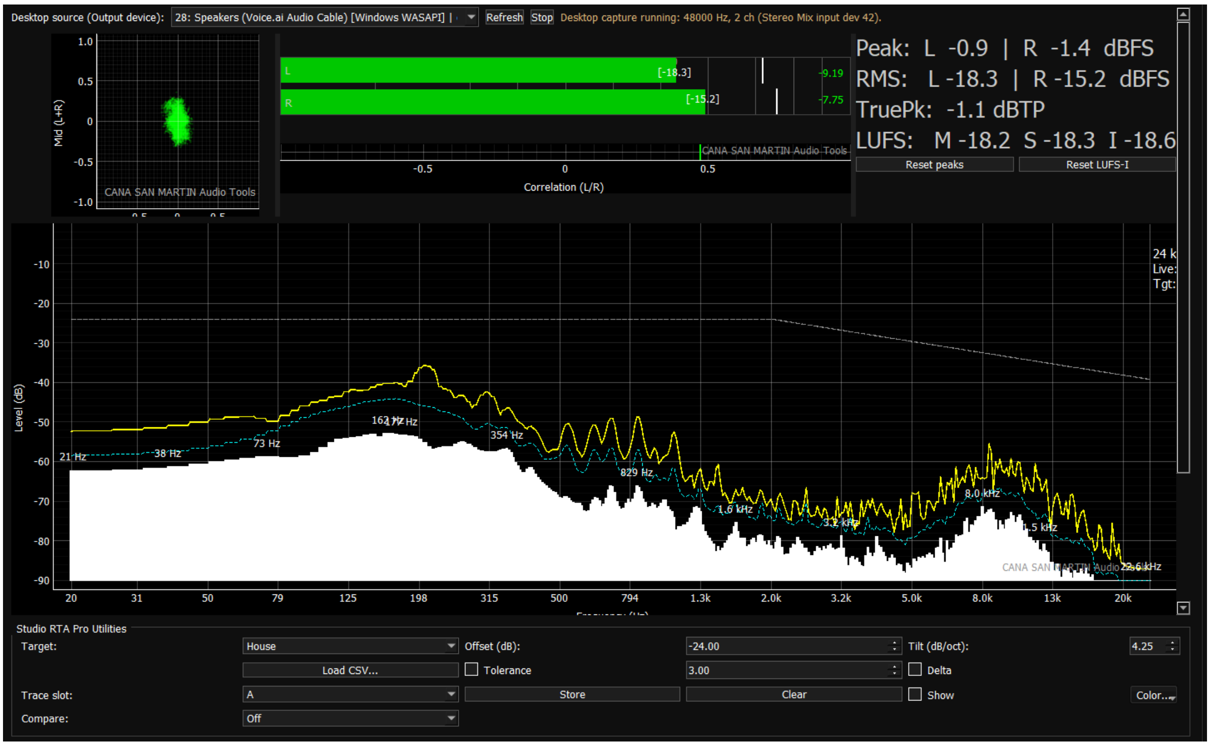
9.8.10 Practical recommendations (to make Wizard 6 "show-stopping")

1. Maintain **the same gain** and routing throughout the entire capture sequence.
2. Capture **representative positions** , not extremes (walls/corners) unless that is the objective.
3. Use reasonable minimum consistency (0.4 is an excellent starting point).

4. Save and name the result: for example
    a. QuickAvg(6) FOH
    b. QuickAvg(6) Audience
    so that the PDF and comparisons then make sense.

10 - Studio monitor tab (Loopback)

This tab is designed to **measure what the computer plays back** (desktop audio) **without a microphone** , that is, *in the digital domain* and in real time. The typical purpose is "mixing/master control" (studios) or deliverable verification (streaming/broadcast), measuring spectrum, stereo/phase, and levels/loudness of the audio output from the selected device.



10.1 What exactly does it measure and why is it useful?

The signal arriving at the chosen output device (speakers/headphones/virtual cable) is captured **before** being converted to an audio signal. On Windows, this is done using **WASAPI loopback** (capturing the "render endpoint"). The tab itself states this as an operating note: it uses loopback and suggests capturing 2 channels even though only 1 channel will be viewed internally.

1. It allows you to analyze **what is actually coming out of the system** (DAW, player, browser, etc.), without errors due to microphone positioning, room, etc.
2. It is ideal for:

- **Spectral balance** (RTA) without acoustic influence.
- **Mono/phase compatibility** (correlation and goniometry).
- **Levels** (Peak/RMS/True Peak) and **loudness (LUFS)** as final delivery control.

Important: As this is a digital capture, the readings are primarily expressed in **dBFS / dBTP / LUFS** . It is not an "on-air" SPL (dB SPL) measurement.

10.2 Loopback capture: operational theory and considerations

HOME Loopback (Windows)

In this implementation, capture is performed by opening an `InputStream` with WASAPI configuration in loopback mode (when available). It verifies that the environment is Windows and that `sounddevice` is exposing `WasapiSettings` , `and loopback=True` is enabled . This is why it may not work correctly in versions prior to Windows 11.

Practical implications:

- **The same output device** through which the audio is played must be selected as the source .
- If the stream doesn't start, it could be due to **channel** , **sample rate** , or driver incompatibility. The tab explicitly suggests "trying another WASAPI device or matching the sample rate" when it fails to start.

10.3 Upper controls: source selection and startup

At the top of the tab is the main control row:

- **Desktop source (Output device):** selector of the output device to monitor.
- **Refresh:** Re-enumerates devices.
- **Start/Stop:** starts/stops the capture.
- **Status:** status/error messages (amber-colored warnings).

Recommended use:

1. Choose the correct device (the same one that is "ringing").
2. Press **Start** .
3. Verify that the **status** indicates operation and that there is movement in meters/graphs.

10. 4 Tab Modules: What each one shows and how to interpret it

The tab integrates a set of typical mix/master control analyzers.

A) VU / PPM (dynamic level measurement)

What it shows

- Signal level bars per channel (depending on the design, may show L/R).
- **It combines RMS** (average energy) and **peak (instantaneous peaks)** concepts with its ballistics.
- **RMS** correlates best with "sense of volume" (sustained energy).
- **Peak** indicates rapid peaks relevant to clipping and headroom.

B) Estadísticas (Peak/RMS/True Peak/LUFS)

The upper right box shows:

- Numerical values of level and loudness (e.g. Peak, RMS, True Peak and LUFS in their variants).
- Reset buttons to "clean" maximums or integrations, depending on version/configuration.
- **True Peak (dBTP):** estimates *inter-sample peaks* (peaks that may appear when reconstructing the analog signal).
- **LUFS:** normalized perceptual loudness (according to windows/times: momentary, short-term, integrated, etc., depending on the module).

C) Goniometer (vector scope / stereo analyzer)

XY cloud/trace measurement that describes the relationship between channels. Can operate in **L/R** or **M/S** depending on the selector.

Essential interpretation:

- **Mono coherent (L≈R, in phase):** tends towards a **vertical line** (maximum positive correlation).
- **Very "wide" signal with strong Side content:** cloud more "open" horizontally.
- **Phase problems / cancellations:** low or negative correlation, cloud tending to horizontal or strange patterns.

Just M/S

1. Mid and Side are linear combinations:
    a. **M = (L + R) / 2**
    b. **S = (L − R) / 2**
2. It is very useful for diagnosing whether the stereo issue originates from truly "side" content or from problematic phase shifts.

D) Correlation meter (degree of correlation L/R)

Indicator (from -1 to +1 pi radians) that quantifies how similar L and R are.

Interpretation

- **+1:** L and R identical (perfect mono).
- **0:** uncorrelated (may be highly uncorrelated stereo or independent noise).
- **-1:** total investment (high risk of cancellation when adding mono).

This module is the quick "thermometer" of mono compatibility.

E) RTA (Real-Time Analyzer) del loopback

The RTA on this tab is **exclusively** for audio captured from the desktop. It exposes:

1. Real-time spectrum (magnitude vs frequency), useful for:
    a. balance tonal,
    b. resonances,
    c. excess/deficiency of bands (subbass, presence, air),
    d. comparison against targets and traces (when enabled).

Main parameters of the RTA (controls)

The tab includes a set of configurable controls within a scrollable area, allowing for expansion without breaking the layout. These controls include:

- **Studio RTA res (fraction of an octave)** : 1/1, 1/3, … 1/48.
- **Integration** : integration time window (125 ms…2 s).
- **Weight** : Flat/A/C.
- **Smooth** : None, 1/24, 1/12, 1/6.
- **RTA ch**: Mid/Left/Right/Side.
- **Gain (dB)** : Display gain.
- **Range (dBFS)** : minimum/maximum of the vertical axis.

FFT size and Window (critical "DSP" parameters)

To parameterize the spectral analysis, the Studio monitor incorporates controls for:

- **FFT size** (512…16384)
- **Window** (Hann, Hamming, Blackman, Rectangular)

These parameters directly affect:

- **Frequency resolution** (large FFT = more resolution, more time latency).
- **Spectral leakage** (the window controls how energy "leaks" between bins; Rectangular has more leakage; Blackman reduces leakage but widens the main lobe, etc.).

When the user changes FFT/window, the system **recreates the DSP core** with the new configuration and updates bands/centers if applicable.

10.5 "Studio RTA Pro Utilities": Targets and Comparison

The tab includes utilities to compare measurements against targets and enable tolerance/delta tools:

1. **Target:** Off / Flat / Tilt / House / Custom CSV.
2. Target parameters:

    a. **Offset (dB)**
    b. **Tilt (dB/oct)**

3. **Load CSV...** to load an external target curve.
4. **Tolerance** and **Delta** to visualize margin and/or difference.

In practical terms, this transforms the loopback RTA into a reference tool comparable to a typical tonal adjustment "target curve".

10.6 Recommended procedure (study workflow) for using Loopback

1. **Configure the audio system**
    a. Select the correct device in “Desktop source”.
    b. Confirm that the audio you want to measure is actually coming through that endpoint.
2. **Start capture**
    a. Click **Start**.
    b. Confirm "running" status and movement on meters.
3. **Verify stereo integrity**
    a. Check **Correlation** : ideally close to +1 for mono material or with high compatibility.
    b. Check **Goniometer** : mono should focus vertically; correct stereo material should form a coherent cloud without inversion patterns.
4. **Analyze spectrum**
    a. Adjust **Studio RTA res** , **Integration** , **Smooth** according to the objective:
        i. Rapid diagnosis: short integration + moderate smoothing.
        ii. fine tonal evaluation: greater integration + high res (e.g. 1/24–1/48) + large FFT.
5. **Compare against target**
    a. Activate **House/Tilt** or load CSV if searching for a defined reference.
6. **Level control**
    a. Monitor Peak/RMS/True Peak and LUFS for delivery consistency (and use resets when appropriate).

10.7 Important limitations and warnings

- **It does not replace acoustic measurement** : it is a "digital" measurement. To evaluate room, SPL, system response in space, etc., a microphone and the other tabs (RTA/TF/Room Parameters, etc.) are required.
- It can capture **everything that sounds through the system** , including notifications, etc., if they share the same output device.
- If the stream does not start, the module itself indicates that another WASAPI device should be tried or the sample rate should be matched.
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