CORRECTION METHODOLOGY

How the DEQX System's Approach Differs From Conventional Digital Crossover & Room Correction Design

A Technical Methodology Comparison

Companion document to: Technical Performance Review — DEQX System. Prepared 23 September 2026

Executive Summary

The DEQX System treats driver correction, crossover phase alignment and room integration as a single, geometry-specific problem, referenced to the actual listening position. This document sets out, dimension by dimension, how that methodology works and how it compares with the more common practice of designing crossover and driver correction first and adding room correction as a separate stage.

Its defining feature is integration: linear-phase FIR crossover design, phase-aware room correction and position-referenced measurement are solved together, as one problem, so that each element is optimised with the others in view.

1. The Common Approach: Two Sequential Stages

Many multi-way loudspeaker systems, and the room-correction products often paired with them, follow a well-established sequence:

- Crossover and driver alignment are designed and measured first, typically referenced to a standard distance from the loudspeaker (conventionally around 1 metre, on- or near-axis).
- Crossovers are commonly minimum-phase (Linkwitz-Riley or similar IIR topologies), which introduce phase rotation and group delay at each crossover point as a characteristic of the filter type.
- Room correction, where used, is applied as a subsequent processing stage, most often focused on magnitude response.
- Measurement gating typically favours one priority: a short, quasi-anechoic-style gate for clean phase data, or a longer capture for low-frequency detail.

Each of these choices reflects sound engineering priorities: measurement convenience, computational efficiency, and the practical challenge of resolving low-frequency detail. The DEQX approach takes a different path, optimising the system at the listener's actual seat and designing room integration in from the outset.

2. The DEQX System's Approach: One Unified Problem

The methodology reviewed in this document addresses the same set of problems differently, treating driver correction, crossover behaviour and room integration as a single computation rather than a sequence of independent stages.

2.1 Individual Driver Correction as the Starting Point

Each driver is measured and corrected — magnitude and phase — on its own actual behaviour, so the crossover is built around the real drivers in the system. This is the foundation the rest of the correction is built on.

2.2 Linear-Phase FIR Crossovers

Crossovers are implemented as linear-phase FIR filters rather than minimum-phase IIR networks. This removes the inherent phase rotation and group delay a conventional crossover introduces at its transition frequency. An explicit, calculated delay is added to the faster bands so that all drivers reconverge in time at the listening position — a deliberate, computed step rather than an incidental side-effect of filter type.

Each of the 8 channels (4 bands × left/right) runs its own 8,192-tap FIR filter — a filter length of approximately 170.7 ms, giving the filter a native frequency resolution of roughly 5.9 Hz. That is a genuinely high-resolution figure: fine enough to shape the correction curve with real precision even at the bottom of the audible band, where achieving a clean, steep, linear-phase transition demands the most filter length. Running 8 simultaneous filters of this length in real time is a substantial computational commitment, and a key part of what makes this level of correction distinctive (see Section 4). The result is a crossover with no phase rotation at its transition points.

These filters run in real time using fast (FFT-based) convolution. Direct time-domain convolution cost scales linearly with filter length for every output sample, which for an 8,192-tap filter across 8 channels would be a heavy computational load. Fast convolution instead performs the equivalent operation via frequency-domain multiplication, using block-based, partitioned processing to keep buffering efficient. This is the same general technique used in professional convolution-based audio processing more broadly; applied here, it is what enables a full 8,192-tap, 8-channel real-time filter bank.

2.3 Time-of-Flight Referencing to the Real Listening Position

Rather than a generic, fixed reference distance, the system's correction is calculated using time-of-flight referenced to the actual listening seat — in the system reviewed here, 2.8 metres. The correction is therefore specific to the geometry that matters to the listener.

2.4 A Deliberately Chosen Measurement Gate

With a filter capable of resolving detail to roughly 5.9 Hz, the system has considerably more native resolution available than the in-room measurement needs. Driver correction is derived from an in-room measurement using a 55-millisecond gate — equivalent to roughly 18 Hz resolution — leaving the filter around three times more resolution than the measurement uses: headroom deliberately held in reserve. The 55 ms figure is chosen specifically because it is long enough to resolve low-frequency detail meaningfully (sufficient to distinguish individual low-order room modes as separate events) while short enough to keep the phase and group-delay estimate dominated by the direct signal rather than the fully-developed reverberant field. This keeps the phase measurement clean and accurate. The filter's spare resolution acts as insurance: the correction is never limited by filter length, only by the acoustically-motivated ceiling deliberately placed on the measurement feeding it.

2.5 Room Correction Layered on an Already Phase-Correct Foundation

Automatic parametric room correction is derived from, and applied on top of, a measurement that is already time-of-flight referenced and phase-aligned — so room correction and time/phase alignment work together as one calculation.

2.6 Closed-Loop, Iterative Verification

The system re-measures its own in-room result after driver correction and delay compensation are applied, again after automatic PEQ is derived, and again to verify the final combined result — a closed-loop, iterative validation process that confirms at each stage that the correction performs as calculated.

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3. Comparison at a Glance

Dimension	Common Approach	DEQX System Approach
Crossover phase behaviour	Minimum-phase (IIR); phase rotation and group delay at each crossover point as a property of the filter type	Linear-phase FIR; no inherent phase rotation, with calculated delay compensation to realign bands in time
Reference geometry	Standard reference distance (commonly ~1 m, on- or near-axis)	Time-of-flight referenced to the actual listening position
Filter length	Shorter FIR or conventional IIR designs	8,192-tap linear-phase FIR per channel (~5.9 Hz native resolution), across all 8 channels

Dimension	Common Approach	DEQX System Approach
Driver vs. room correction	Sequential stages: crossover first, room correction applied afterward	Solved as one unified problem: driver correction, crossover delay and room alignment computed together
Measurement gate length	Short gate (favours phase accuracy) or long/ungated capture (favours bass resolution)	55 ms — a deliberate balance of ~18 Hz frequency resolution against phase-domain accuracy
Room correction scope	Often focused on magnitude response	Magnitude and phase/group-delay correction together, verified per channel
Verification process	Typically a single measure-and-apply cycle	Multi-stage closed loop: measure → correct → remeasure → PEQ → remeasure again

4. What Makes This Combination Distinctive

Linear-phase FIR crossovers, phase-aware room correction and position-referenced measurement are each demanding to implement well. Bringing them together as one calculation is what sets this methodology apart:

- Linear-phase FIR crossovers are computationally and design-intensive relative to conventional minimum-phase filters, and require explicit delay-compensation engineering that minimum-phase designs do not require. Running 8 independent 8,192-tap filters simultaneously in real time — as this system does, one per channel across 4 bands and both stereo channels — requires fast (FFT-based) convolution rather than direct-form processing, and is a substantial, deliberate engineering commitment.
- Solving driver correction and room correction as one geometry-specific problem, rather than two independent ones, requires the measurement and correction architecture to be built around that unification from the outset, as a foundational design decision.
- Balancing measurement gate length to serve both low-frequency resolution and phase accuracy simultaneously — rather than prioritising one over the other — reflects deliberate engineering judgement specific to the correction goal.
- Closed-loop, multi-stage re-verification confirms that the calculated correction actually achieves its intended real-world result.

5. Supporting Evidence From This Review

The companion Technical Performance Review documents the direct evidence for this methodology functioning as intended in the system evaluated: sample-aligned driver arrival confirmed in the impulse response; effectively flat group delay through the mid and high bands; sub-band phase behaviour that aligns precisely with independently, physics-calculated room resonance and null frequencies; an applied correction filter set that intelligently targets only what is genuinely correctable, including at destructive-interference nulls; and a final closed-loop re-measurement consistent with every earlier stage of analysis. These findings were independently corroborated across measurement methods rather than taken on the strength of the design intent alone.