TELPA AUDIO

ENGINEERING WHITE PAPER

Active Studio Sound Monitor ASM-8RS

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Abstract

In modern critical listening environments, conventional loudspeaker evaluation methodologies that rely primarily on single-frequency swept sines (THD) and idealized steady-state frequency response measurements fail to predict system behavior under complex dynamic broadband excitation. Real-world program material consists of dozens of simultaneous spectral components interacting within voice coils, magnetic motor structures, and amplification stages.

This technical paper presents the electroacoustic principles and objective verification data of the Telpa Audio ASM-8RS active studio sound monitor. It details the physical mechanisms governing multitone intermodulation distortion (MTIM) suppression, digital linear-phase FIR filtering for near-zero group delay, the acoustic suspension dynamics of an unvented enclosure, and per-unit individual FIR calibration ensuring a guaranteed production batch tracking corridor of ±0.8 dB.

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1. Architectural Overview & Acoustic Topology

1.1 Transducer Integration & Acoustic Loading

The electroacoustic topology integrates:

- Low-Frequency Transducer: Custom 200 mm (8-inch) long-throw mid-bass driver with a 51 mm (2-inch) voice coil, 200 W power handling, and 93 dB nominal sensitivity.



Fig. 1 Front view of ASM-8RS.

- High-Frequency Transducer: 44 mm (1.75-inch) voice coil neodymium compression driver (40 W power handling, 110 dB sensitivity) coupled to a constant-directivity horn flare (1" throat diameter, 90° H× 60°V coverage).

- Hybrid Bi-Amplification: Dedicated power architecture pairing a 250W RMS Class-D amplifier stage for low-frequency transient speed paired with a discrete Class-A/AB amplifier stage dedicated to the high-frequency compression driver, operating in pure symmetrical Class-A up to 97 dB SPL for zero crossover distortion and natural harmonic decay, and transitioning seamlessly to Class-AB for peak dynamic headroom.

- DSP Core: 32-bit processing at a native 96 kHz internal resolution executing driver protection, room target profiling, crossover alignment at 500 Hz, and linear-phase FIR phase correction.

1.2 Low-Frequency Acoustic Alignment & Dynamic Headroom

The ASM-8RS utilizes an acoustic suspension topology configured within a rigid, low-resonance enclosure. The primary acoustic chamber features an internal gross volume of 22 L. Accounting for driver assemblies, internal perimeter window bracing, and horn flare displacement, the clear internal air volume is about 18 L. The cavity is filled with low-density acoustic damping material, yielding an effective acoustic operating volume of approximately 20 L via virtual compliance expansion.

Coupled to the custom 8-inch low-frequency transducer (f_s = 64 Hz, V_{as} = 26 L, Q_{ts} = 0.31, X_{max} = ± 7mm), this acoustic air spring establishes an in-box system resonance $f_c \approx 96$ Hz with an alignment factor of $Q_{tc} \approx 0.47$.

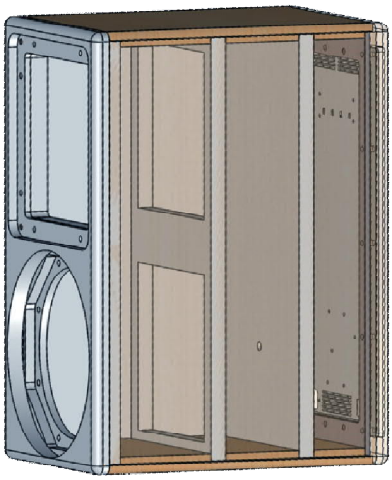


Fig. 2 ASM-8RS Enclosure cutaway.

Operating below the classical Bessel threshold (Q = 0.50), this critically damped alignment prevents resonant ringing and overhang, ensuring optimal time-domain decay and minimal group delay down to the low-frequency cutoff.

To deliver full-bandwidth monitoring in factory reference FLAT mode, the unassisted 12 dB/octave mechanical roll-off is linearized via precision DSP filtering down to the −3 dB half-power boundary at 32 Hz. Rather than forcing an artificial flat extension down to 20 Hz—which would induce severe intermodulation distortion and power compression—the system profile follows a controlled roll-off to −9 dB at 20 Hz.

Even under high-level program transients with a +10 dB crest factor (95 dB SPL at 1 m), peak cone excursion remains capped at approximately 3.5 mm. This maintains driver operation well within its linear magnetic gap, preserving dynamic headroom and ensuring clean, low-distortion sub-bass reproduction.

2. Group Delay & Linear Phase Alignment

2.1 The Time-Domain Problem

Standard analog crossovers and vented enclosures introduce cumulative phase shifts and frequency-dependent group delay. In bass-reflex systems, phase rotation near the port resonance frequency causes time-domain ringing and transient smearing, which reduces pitch clarity in lower registers and degrades phantom center imaging.

2.2 FIR Phase Linearization in the ASM-8RS

The ASM-8RS incorporates digital finite impulse response (FIR) phase linearization algorithms to achieve negligible group delay (<0.5 ms) across the entire audible spectrum from 80 Hz up to 40 kHz.

Below 50 Hz, group delay transitions into the smooth, gradual curve dictated by the physical minimum-phase behavior of sealed enclosures (12 dB/octave roll-off), avoiding the steep 24 dB/octave phase rotation and resonant overhang inherent to ported cabinets.

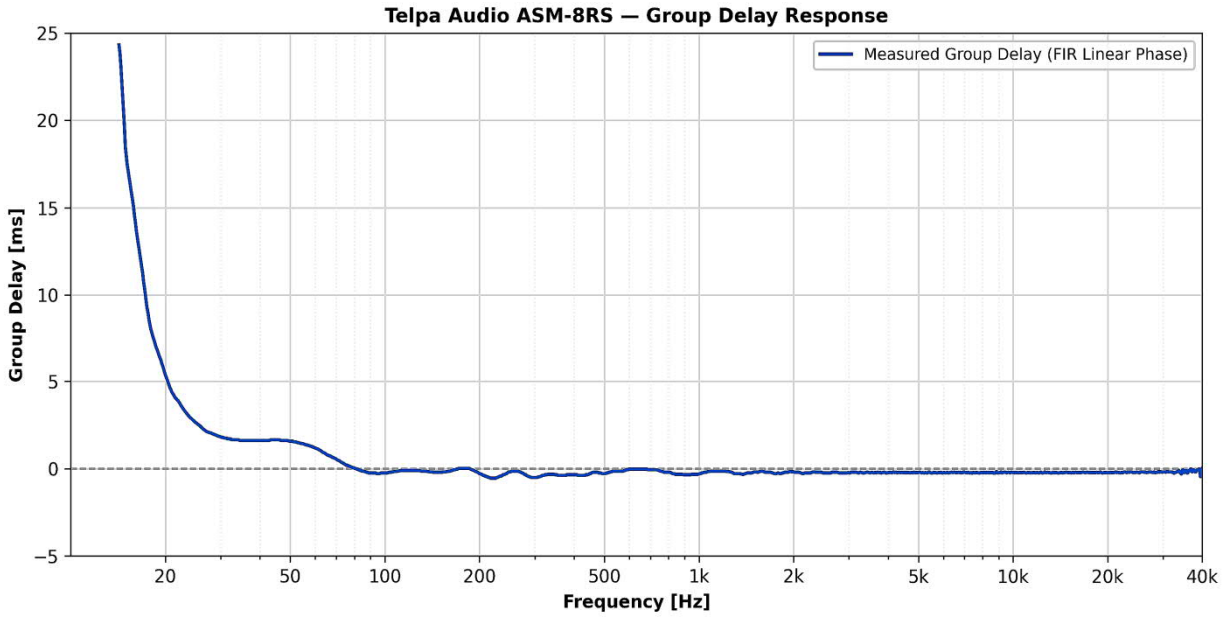


Fig. 3 Measured Group Delay Response with FIR Linear-Phase Correction.

System Latency: The processing engine maintains an end-to-end latency of ~4 ms in full linear-phase mode, preserving real-time operational capability during tracking and live instrument recording.

3. Multitone Intermodulation Distortion (MTIM)

3.1 Harmonic vs. Intermodulation Distortion

While Total Harmonic Distortion (THD) measurements are useful for diagnosing mechanical rub or structural resonances, single swept sines evaluate transducers under static conditions. Musical signals do not consist of isolated tones. When broadband program material passes through non-linear magnetic fields (flux modulation), varying voice-coil inductance L(x), and mechanical suspension stiffness asymmetries $K_{ms}(x)$, multiple frequencies intermodulate.

- Harmonic Distortion: Produces integer multiples ($2f_1$, $3f_1$), which are naturally masked by musical overtones.

- Intermodulation Distortion (IMD): Generates non-harmonic sum and difference products ($f_1 \pm f_2$, $2f_1 \pm f_2$) that have no musical relationship to the program material. In human perception, intermodulation is directly recognized as acoustic congestion, harshness, loss of depth, and blurred instrument separation during dense orchestral, rock, or electronic passages.

3.2 MTIM Evaluation Methodology

The Multitone Intermodulation test utilizes a broadband stimulus comprising multiple logarithmically spaced, non-harmonically related sine waves covering the complete audible bandwidth. By intentionally spacing excitation tones at non-integer multiples, empty frequency intervals (bins) are created across the spectrum. Acoustic energy measured inside these vacant bins represents pure non-linear artifacts generated by the transducers and amplifier circuitry.

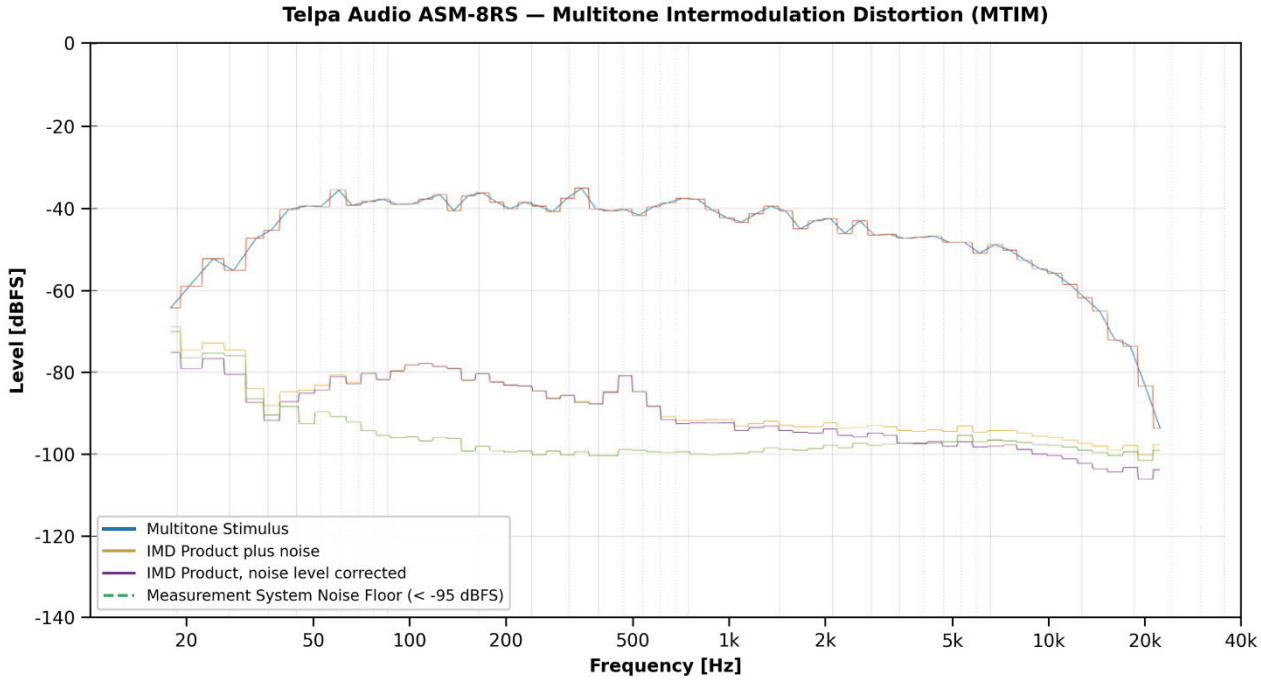


Fig. 4 Multitone Intermodulation Distortion (MTIM) Spectrum with Noise-Floor Correction.

3.3 Quantitative MTIM Analysis

With measurement system background noise mathematically subtracted (noise-level corrected trace), the ASM-8RS maintains a signal-to-distortion separation exceeding 45 dB to 55 dB across the primary operating range, rising beyond 55 dB in the high-frequency band. Spurious intermodulation artifacts remain suppressed below 0.2% to 0.5% under high-level broadband excitation:

Table 1: Measured Multitone Intermodulation Distortion (MTIM) Performance

Frequency Range	Signal-to-IMD Separation (dB)	IMD Artifact Level (dBFS)	Equivalent Linear Distortion (%)	Acoustic Impact & Performance
30 Hz – 100 Hz	42 – 48	-80 to -85	< 0.8%	Excellent woofer control in sealed enclosure; maintains clean bass pitch differentiation without mid-bass mud.
100 Hz – 500 Hz	46 – 52	-82 to -88	< 0.5%	Minimal voice-coil flux modulation; preserves body and clarity in kick drums, bass guitar, and lower vocals.
500 Hz – 2 kHz	50 – 54	-92 to -94	< 0.3%	Seamless crossover transition; prevents vocal harshness and maintains transparent center-image placement.
2 kHz – 10 kHz	52 – 58	-94 to -100	< 0.25%	Ultra-low compression driver intermodulation; high spatial micro-detail, clear cymbal shimmer, and transient air.
10 kHz – 20 kHz	54 – >60	-98 to -106	< 0.2%	Compression driver operates well within linear displacement; pure harmonic decay with no smearing or fatigue.

4. Amplitude Frequency Response & Production Consistency

4.1 Frequency Bandwidth

In factory reference FLAT mode, the ASM-8RS delivers an exceptionally linear amplitude response from 45 Hz to 22 kHz, adhering strictly to a ±0.8 dB production corridor with core passband deviation within ±0.2 dB.

The electroacoustic alignment provides an extended low-frequency cutoff of 32 Hz (−3 dB) and usable sub-bass reach to 20 Hz (−9 dB) via optimal acoustic suspension damping. At high frequencies, the system maintains passband linearity up to 22 kHz before entering a smooth, constant-directivity acoustic roll-off that reaches −3 dB at 26 kHz.

The custom horn flare geometry guarantees uniform dispersion and controlled directivity across the entire treble band, free from parasitic diaphragm breakup modes in the audible spectrum.

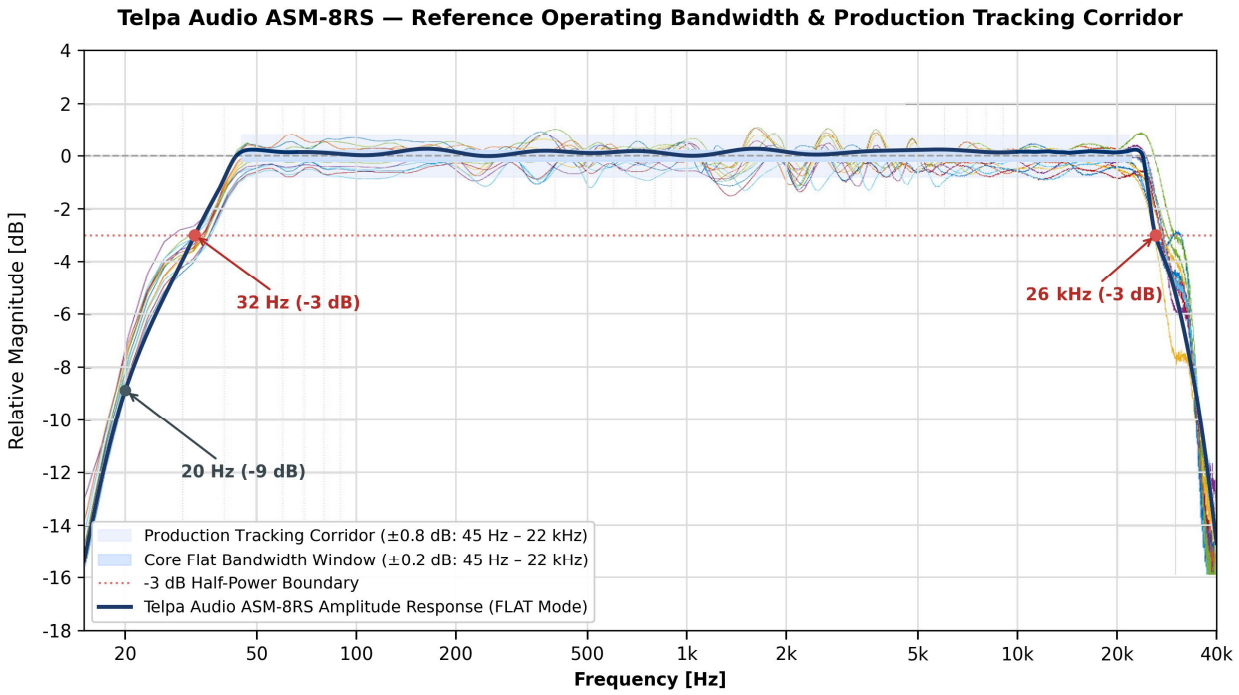


Fig. 5 On-Axis Free-Field Frequency Response (FLAT Mode) with Production Tracking Corridor and Boundary Cutoffs.

4.2 Individual Unit Calibration & Stereo Pairing

Manufacturing tolerances in driver mechanical suspensions and magnetic assemblies inevitably introduce deviations between individual speakers, distorting the stereo soundstage and causing phantom center drift.

To eliminate unit mismatch, every single ASM-8RS monitor undergoes individual acoustic measurement during final assembly. Dedicated correction coefficients are generated and loaded directly into the internal DSP to match the unit's transfer function to the golden laboratory reference.

- Guaranteed Tracking Corridor: All production units track within a verified ±0.8 dB envelope from 45 Hz to 22 kHz.
- Acoustically Matched Pairs: Any two randomly selected ASM-8RS monitors form a precision-matched stereo pair, regardless of manufacturing date or serial number sequence.

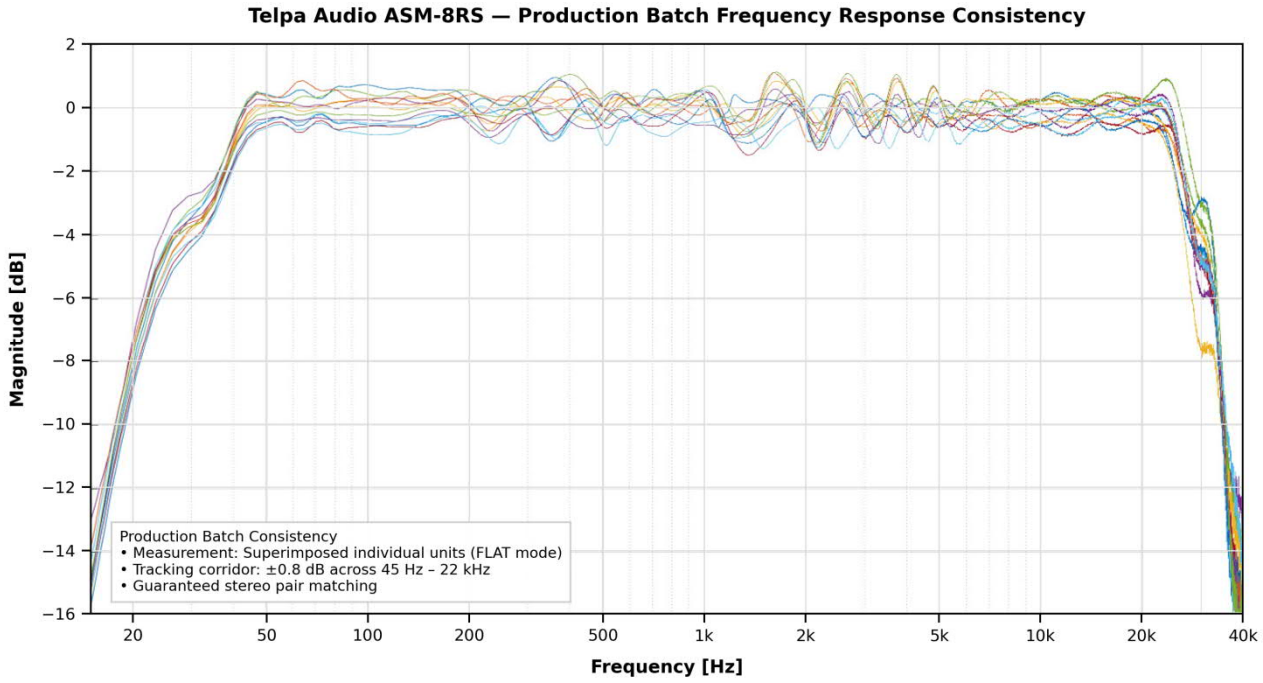


Fig. 6 Measured Amplitude Frequency Response (AFR) and Production Batch Consistency across Serial Production Units (FLAT Reference Mode).

5. Acoustic Adaptation & Protection Topologies

Three boundary compensation profiles and a switchable subsonic protection filter are embedded in DSP:

- FLAT Reference Mode: Factory baseline calibration delivering a phase-linearized FIR response across the full frequency range. Recommended for acoustically treated control rooms with minimal boundary interference.
- Preset 2 (7 dB Room Tilt): Provides a continuous acoustic tilt anchored at 0 dB at 1 kHz, transitioning to a +3.5 dB low shelf below 31.25 Hz and rolling off to −3.5 dB at 20 kHz. Compensates for room reflection characteristics in standard control rooms.
- Preset 1 (9 dB Room Tilt): A steeper downward contour with +4.5 dB low shelf below 31.25 Hz and −4.5 dB at 20 kHz. Recommended for untreated or reflective control environments where high-frequency buildup must be mitigated.
- LF CUT (40 Hz Protection Filter): A 2nd-order active high-pass filter (12 dB/octave, −3 dB at 40 Hz). Provides >15 dB attenuation at 20 Hz and >27 dB at 10 Hz. Prevents woofer over-excursion and protects dynamic amplifier headroom during elevated monitoring SPL.
- Combined Modes (Dotted Lines): Engaging LF CUT ON while running either Preset 1 or Preset 2 maintains the desired tonal contour across the midrange and high frequencies while cleanly cutting sub-40 Hz energy.

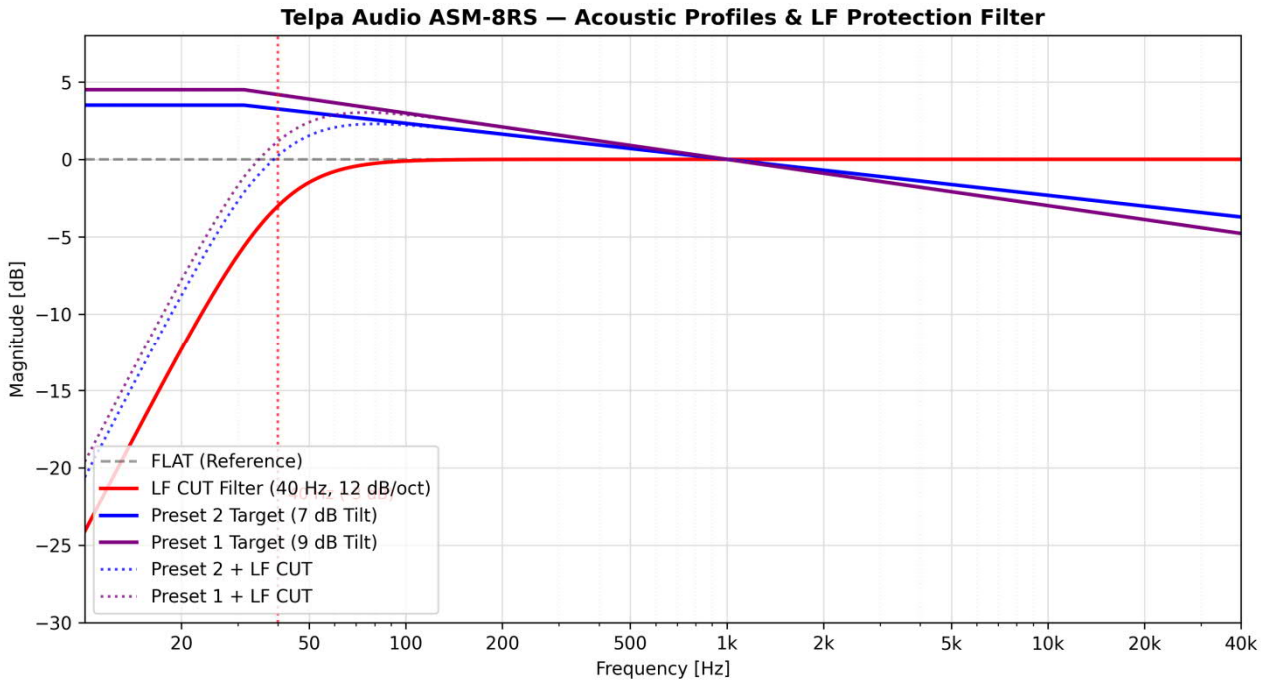


Fig. 7 Acoustic Profile Target Curves and LF CUT 40 Hz Protection Filter Response.

6. Technical Specifications

Parameter	Specification
System Classification	2-Way Active Studio Monitor, Sealed Acoustic Suspension
Low-Frequency Transducer	200 mm (8-inch) mid-bass driver, 51 mm (2-inch) voice coil, 200 W power handling, 93 dB sensitivity
High-Frequency Transducer	44 mm (1.75-inch) voice coil neodymium compression driver, 40 W power handling, 110 dB sensitivity
Horn Flare / HF Dispersion	Constant-directivity horn (1-inch throat diameter); 90° H × 60° V coverage
Frequency Response (FLAT)	45 Hz – 22 kHz (±0.2 dB)
Linear Bandwidth Extension	32 Hz – 26 kHz (-3 dB)
Batch & Stereo Pair Tracking Corridor	±0.8 dB (45 Hz – 22 kHz)
Maximum Continuous SPL	108 dB SPL @ 1 m
Maximum Peak SPL	120 dB SPL @ 1 m (pair)
Amplification Topology	Hybrid Bi-amplification (LF - 250 W RMS Class-D / HF - Class-A (>97dB -AB))
DSP Architecture	32-bit, 96 kHz internal resolution
Filter Topology	Linear-phase FIR correction & crossover alignment, 500 Hz crossover frequency
System Latency	~4 ms (Linear-phase mode)
Acoustic Profiling	FLAT (Reference), Preset 1 (-9 dB Tilt), Preset 2 (-7 dB Tilt)
Protection High-Pass Filter	LF CUT: 40 Hz, 12 dB/octave 2nd-order Butterworth response
Analog Input	Balanced 3-pin XLR female (+4 dBu nominal, max input level +24 dBu, 10 kΩ impedance)
Digital Input	AES3 (AES/EBU) 110 Ω balanced XLR female, 44.1 kHz to 192 kHz sample rates
A/D & D/A Conversion	24-bit / 192 kHz high-dynamic-range converters
Operating Mains Voltage	Universal AC 100–240 V, ~50/60 Hz
Power Consumption	Idle: 18 W; Maximum Output: 400 W
Dimensions (H × W × D)	470 mm × 260 mm × 360 mm
Net Weight	18.0 kg