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Research on Testing Methods for Intelligent In-Vehicle Audio System

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Abstract

With the advancement of automotive intelligence, the acoustic environment within the cabin has become a critical dimension of user experience. This paper addresses the evaluation needs of intelligent cabin audio systems by proposing a comprehensive testing and assessment methodology based on user experience. Initially, the functional and performance metrics of in-vehicle intelligent audio systems are analyzed to define evaluation dimensions, including wake-up capability, interaction performance, distortion rate, sound quality, and active noise cancellation efficiency, with corresponding testing protocols designed accordingly. Subsequently, audio signals are collected under various cabin conditions (such as driver and passenger positions, urban and highway driving scenarios) to quantify system performance through both subjective and objective evaluation methods. Experimental results demonstrate that this approach effectively assesses the performance of intelligent audio systems across multiple metrics. By applying weighted calculations, an overall system score is derived, providing a foundation for optimizing the design of intelligent audio systems. This study bridges a gap in intelligent cabin audio testing and holds significant implications for enhancing in-vehicle acoustic quality and driving comfort.

Keywords

In-Vehicle Audio Effects; Audio Analysis; Vehicle-Level Testing.

1. Introduction

With the rapid advancement of intelligent connected vehicle technology, automobiles are evolving from conventional transport means to intelligent mobile spaces, where smart cabins have become central to human-machine interaction and entertainment experiences [1]. Sound, owing to its immersive and intuitive nature, significantly impacts driving safety and comfort; modern vehicular audio systems utilize multi-speaker configurations and digital signal processing to achieve spatialized playback and immersive auditory experiences [2]. However, in complex in-cabin noise environments, relying solely on traditional structural sound insulation falls short of simultaneously achieving optimal audio quality and quietness, making active noise control an increasingly critical approach [3]. As user expectations for cabin sound experiences rise, the industry requires more systematic evaluation methodologies and comparable metrics to guide product enhancement [4].

At the systemic and methodological level, existing research has proposed Ethernet-based automotive audio architectures enabling multi-source playback and sound field focus management, thereby enhancing flexibility in audio rendering [2]. Other studies have employed finite element methods to acoustically optimize components like doors, improving sound field uniformity and frequency response characteristics in speaker installation zones [5]. For noise control, real-vehicle low-frequency active noise cancellation systems have demonstrated engineering feasibility by employing adaptive algorithms to counteract primary noise sources in real-time [6]. A comprehensive review on active sound quality control synthesizes progress from FELMS to data-driven methodologies, offering methodological insights for refining in-car acoustic environments [3]. Concurrently, standardized objective testing protocols for vehicular noise continue to be refined, with metrics and procedures designed around system characteristics to enhance measurement repeatability and validity [7]. Collectively, significant strides have been made in system architecture, structural acoustics optimization, active noise reduction, and objective testing. Nevertheless, bridging multi-source technical parameters with genuine user perception to establish scientific, systematic, and replicable comprehensive testing methods remains a pivotal challenge requiring resolution.

2. Overview of In-Vehicle Audio System

The in-vehicle voice interaction system captures speech inputs from occupants, extracts command data, and invokes corresponding application services to enable secure and efficient human-vehicle interaction. This system comprises two key components: a low-power front-end processing module and a higher-power speech processing module. Typically, the front-end module remains active while the speech processor stays in standby mode. The speech processing module is activated by the front-end module only when voice interaction is required, thereby optimizing power consumption.

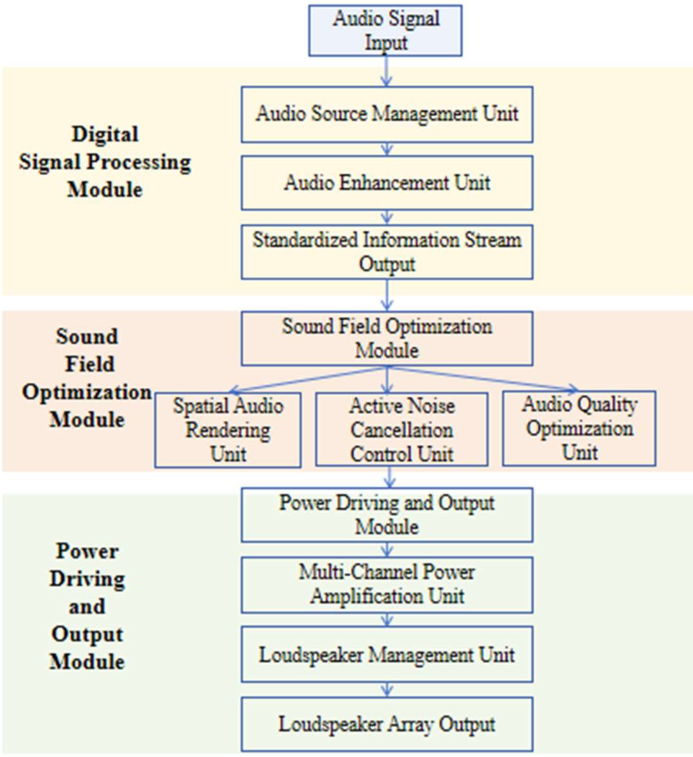

Figure 1. Structure of the In-Vehicle Audio Effect System

2.1. Digital Signal Processing Module

The digital signal processing module comprises an audio source management unit and an audio enhancement unit, primarily responsible for standardizing multi-channel audio inputs through format unification, level matching, and basic sound effect processing to generate standardized signal streams for subsequent sound field reconstruction.

Vehicle audio systems must concurrently process multiple audio sources including navigation prompts, media playback, and Bluetooth calls, which exhibit differences in sampling rates,

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dynamic ranges, and priority levels. The audio source management unit employs routing and mixing strategies to coordinate multi-source playback while preventing informational masking. The audio enhancement unit performs dynamic range control, equalization compensation, and filtering to improve frequency response characteristics and auditory perception of original audio signals.

Within complex vehicular acoustic environments, constraints on speaker placement, cavity resonance, and reflected sound interference significantly degrade playback quality. Pre-correction based on in-cabin acoustic transfer functions is therefore implemented. Predominant solutions utilize digital signal processors (DSP) to achieve parametric equalization, delay compensation, and crossover management. As illustrated in Figure 2, the core processing workflow involves room acoustic compensation, frequency-band energy redistribution, transient response optimization, and dynamic distortion suppression applied to original audio signals.

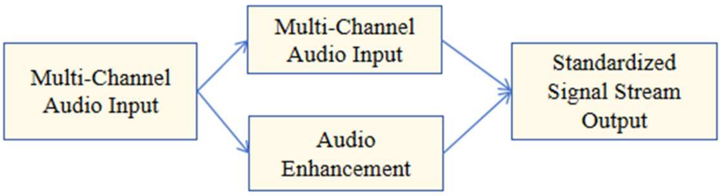

Figure 2. Block Diagram of the Digital Signal Processing Module.

2.2. Sound Field Optimization Module

The sound field optimization module comprises three core components: a spatial audio rendering unit, an active noise control unit, and a sound quality enhancement unit.

Spatial audio rendering utilizes multi-speaker arrays to reconstruct surround sound fields and achieve precise sound localization. Employing head-related transfer functions (HRTF) or surround sound codec technologies, this unit creates coherent sound image distribution and immersive experiences in target listening zones. To address acoustic asymmetries caused by cabin geometry, the system incorporates seat-position awareness to dynamically adjust channel gains and delay parameters, ensuring consistent auditory perception across all seating positions.

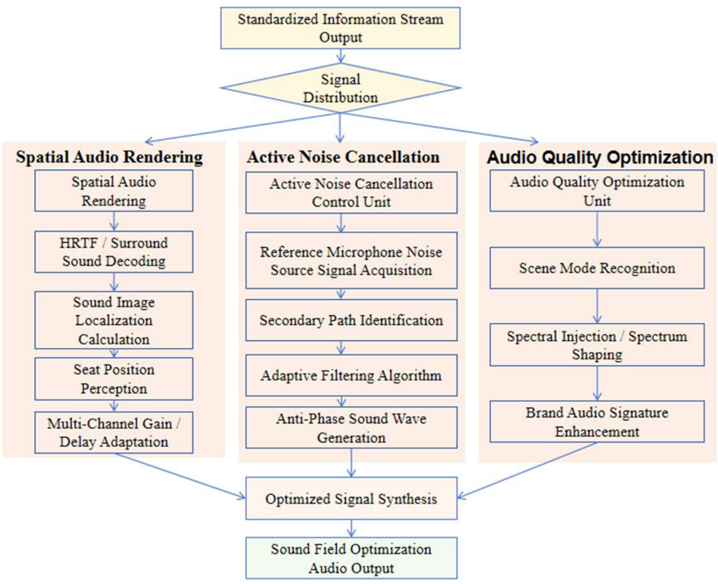

Figure 3. Block Diagram of the Sound Field Optimization Module

Active noise control operates on the principle of secondary sound source synthesis, generating anti-phase waves to cancel low-frequency narrowband noise from powertrain, tire-road contact, and wind sources. Using reference microphones to capture noise signals, the unit drives speakers to emit inverse-phase waveforms through adaptive filtering algorithms, significantly reducing sound pressure levels at the ear. This technology achieves 10-15 dB noise reduction in the 20-500 Hz range, substantially improving speech intelligibility and acoustic comfort.

The sound quality enhancement unit focuses on subjective perception improvement for specific scenarios, such as engine sound augmentation and Acoustic Vehicle Alerting System (AVAS) design for electric vehicles. Through harmonic injection and spectral shaping techniques, it enhances auditory pleasantness and brand identity while maintaining regulatory compliance.

2.3. Power Drive and Output Module

The power drive and output module integrates multichannel power amplification and speaker management units, responsible for converting processed digital signals into acoustic energy output.

The multichannel power amplifier unit employs high-efficiency topologies such as Class D architecture to provide independent power channels for each speaker. This unit incorporates overload protection, temperature monitoring, and impedance detection functions to ensure system reliability under varying operating conditions. Through co-design of DSP and power amplifiers, channel-level independent control is achieved, enabling advanced features including independent audio zones and sound field focusing.

The speaker management unit handles crossover network implementation, driver status monitoring, and fault diagnostics. In systems supporting independent audio zones, this unit can activate distinct speaker combinations for different listening areas. Utilizing beamforming or acoustic field isolation techniques, it achieves physical separation of audio sources between front and rear seating areas, accommodating differentiated listening requirements for drivers and passengers.

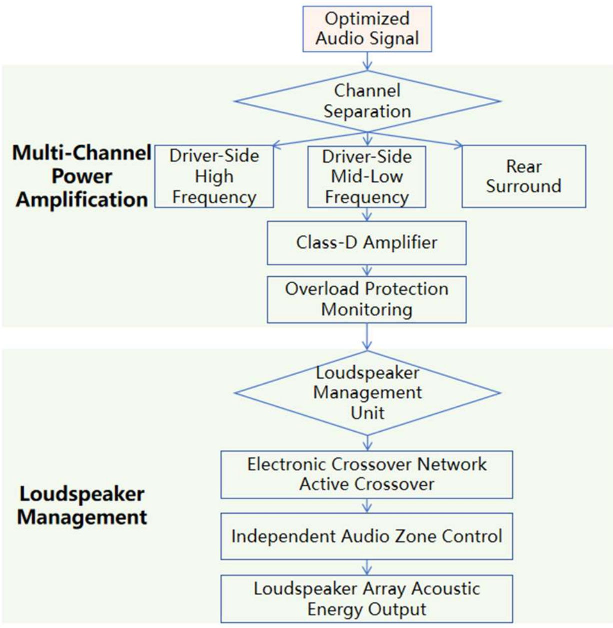

Figure 4. Block Diagram of the Power Driving and Output Module

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3. Intelligent In-Vehicle Audio Testing Technologies and Methodologies

3.1. Overview of the Testing System

During testing, vehicles maintain standardized system volume and sound effect presets. Using standardized test signals (e.g., pink noise, ESS sweeps) as input, combined with multi-speaker configurations and digital processing chains, evaluations are conducted on core parameters including frequency response consistency, left-right amplitude-frequency difference, inter-seat sound pressure level variation, maximum sound pressure at low distortion, and end-to-end latency.

To simulate auditory perception at actual seating positions, measurement apparatus conforming to human head and torso acoustical characteristics (such as HATS) or microphone arrays are deployed. These are positioned at four designated locations-driver, front passenger, and left/right second-row seats-maintaining consistent reference height, horizontal alignment, and headrest distance throughout the testing cycle. The experimental setup configuration (Figure 4) demonstrates: audio interface/amplifiers maintaining standard connections with vehicle head units; HATS (or microphone arrays) aligned and secured relative to seat center points; high-frame-rate cameras facing center console displays to comprehensively document testing procedures and result visualization; all device parameters (gain, sampling rate, bit depth) uniformly configured and recorded for traceability.

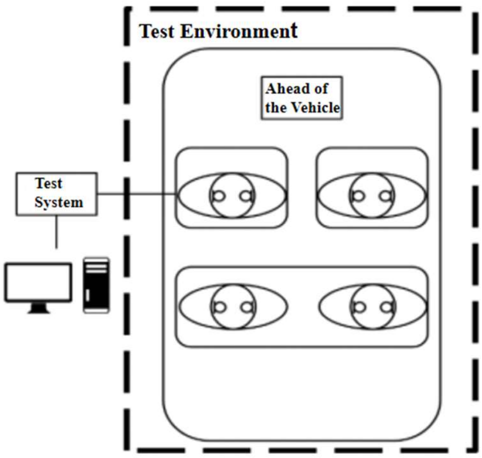

Figure 5. BlockBlock Diagram of the Test System

3.2. Design of Measurement Signal

To ensure comprehensive and consistent testing, measurement signals are categorized into two types with the following specifications:

Signal Type 1 (Pink Noise): Sampling frequency: 48 kHz,Bit depth: 24 bit,Mono format,Peak level: -3 dBFS (crest factor: 12 dB),Frequency range: 20 Hz–20 kHz (without bandpass filtering).Pink noise is employed for distortion and sound quality evaluations. Gain may be adjusted based on actual measurement conditions, provided such adjustments are explicitly documented.

Signal Type 1 (ESS Logarithmic Sweep):Sampling frequency: 48 kHz,Bit depth: 24 bit,Mono format,Standard gain: -15 dB,Sweep range: 20 Hz–20 kHz,Sweep duration: 5 seconds.The combined application of these signals enables systematic frequency response analysis while ensuring high-precision measurements for distortion and spatial consistency assessments.

3.3. Distortion Measurement

To evaluate the system's signal reproduction capability across different occupant positions, a microphone array is used to measure the Total Harmonic Distortion (THD) at the driver's seat, front passenger seat, and second-row passenger seats. The test employs Signal Type 1 at a sampling rate of 48 kHz, with analysis performed in 1/3-octave bands over the 20 Hz to 20 kHz frequency range.The Total Harmonic Distortion is calculated using the following formula:

$$\text{The Total Harmonic Distortion} = \sqrt{\sum_{n=2}^{H}\left(\frac{G_n}{G_1}\right)^2} \tag{1}$$

where $G_n$denotes the RMS value of the n-th harmonic frequency component, and G_1 represents the RMS value of the fundamental frequency component. The parameter H is determined by the sampling frequency; for instance, with a 48 kHz sampling rate, the maximum measurable fundamental frequency is 12 kHz, in which case H is set to 2.

Measurements are conducted separately at three designated positions:

(1)Driver's position: The microphone array is positioned at the center of the headrest area to capture test signals and compute THD values at corresponding frequency points.

(2)Front passenger position: The array is deployed in the front passenger headrest zone using identical measurement methodology.

(3)Second-row right passenger position: The array is arranged in the rear right passenger headrest area to measure and calculate THD values across all frequency points of interest.

3.4. Active Noise Cancellation Efficiency

Active Noise Control (ANC) performance is evaluated under two typical operating conditions-highway and urban driving-to quantify the system's noise attenuation capability across different seating positions

(1) Highway Condition

With ANC deactivated, the vehicle maintains a constant speed of 120 km/h while 1/12-octave sound pressure levels (SPL) across 20 Hz–20 kHz are recorded. The ANC system is then activated, and measurements are repeated at identical speed. The SPL difference (ΔSPL) represents the noise attenuation achieved.

Measurement points include driver, front passenger, and second-row right passenger positions.

(2) Urban Condition

Following the same testing protocol as for highway conditions, the vehicle operates at a steady 60 km/h. Identical measurement points are used.

Comparative analysis of ΔSPL values enables assessment of ANC effectiveness and performance consistency across varying operational scenarios and locations.

3.5. Sound Quality Assessment

Sound quality assessment encompasses three dimensions: fidelity, spatial perception, and energy perception.

(1) Fidelity

Microphone arrays are deployed at the driver's, front passenger's, and second-row right positions to capture Signal Type 1. Measurements are conducted at a 48 kHz sampling rate across the 20 Hz–20 kHz frequency range, with spectral analysis performed in 1-octave bands. Audio reproduction characteristics are evaluated by comparing frequency response curves against standardized thresholds.

(2) Spatial Perception

This dimension is characterized by sound field uniformity and inter-channel amplitude-frequency discrepancy.Sound Field Uniformity: HATS devices simultaneously capture signals at the driver's, front passenger's, and second-row right positions, calculating the maximum A-weighted SPL variation across the 20 Hz–20 kHz range.Left-Right Channel Difference: 1/3-octave responses of left/right channels are analyzed separately at driver and front passenger positions, referenced to the averaged sensitivity at 1 kHz.

(3) Energy Perception

HATS measurements at all positions determine the maximum low-distortion sound pressure level. The audio system incrementally increases volume from minimum level while analyzing THD in 1/12-octave bands (100 Hz–4 kHz) at 48 kHz sampling rate. The A-weighted SPL is recorded as the maximum low-distortion level when all frequency points maintain THD below 10%.

3.6. Independent Sound Zone Testing

HATS systems are positioned at the driver's seat and second-row right seat respectively. With the independent acoustic zone function deactivated, the rear HATS artificial mouth emits test signals while the driver-position HATS artificial ear captures and records 1/12-octave A-weighted SPL values across the 20 Hz–20 kHz range. The independent acoustic zone function is then activated for repeated measurement. The resulting SPL difference constitutes the quantitative metric for acoustic zone isolation performance.

3.7. Adaptive Sweet Spot Testing

The adaptive sweet spot functionality is evaluated under varying occupant configurations through three representative scenarios:1. Single occupant in driver's seat.2. Occupants in driver and front passenger seats.3. All four seating positions occupied

Manual verification determines whether the system automatically optimizes the optimal listening position based on occupant distribution. The functionality is classified as:"Supported" if real-time adaptive optimization is achieved across multiple seats."Partially Supported" if only limited positions demonstrate responsiveness."Not Supported" if no adaptive behavior is observed

4. Validation of Intelligent In-Vehicle Audio Testing Methods

To validate the effectiveness of the testing methodology, this study conducts performance evaluations on the voice interaction system of an electric vehicle model.

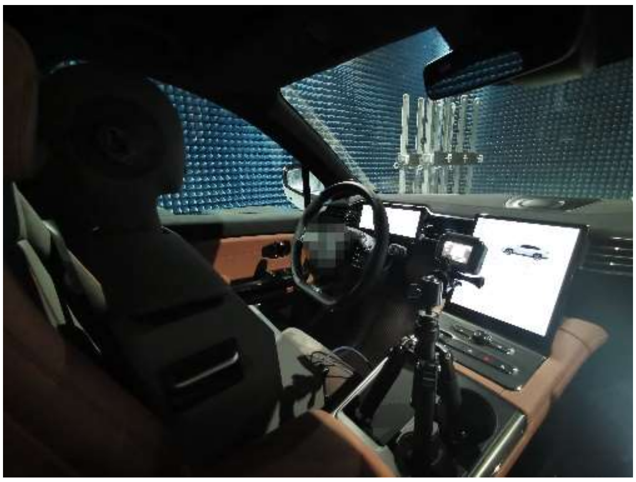

Figure 6. Test Case for Voice Audio System

4.1. Test Conditions

Testing shall be conducted in an acoustically controlled environment where the in-cabin background noise level complies with NC20 specifications, not exceeding -74 dBPa(A). The NC standard reference curve is illustrated in Figure 7.

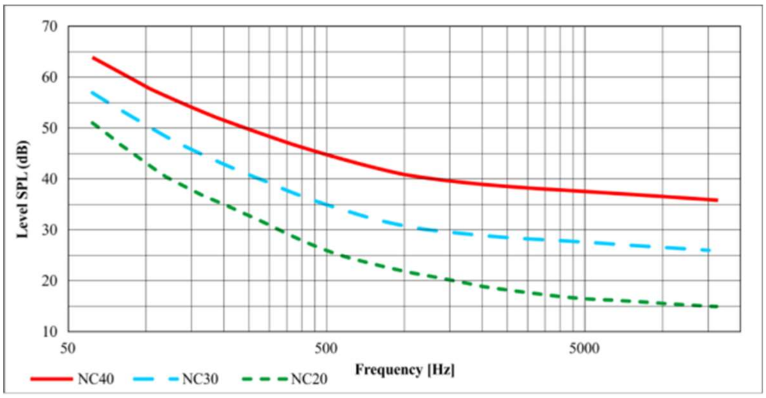

Figure 7. NC Standard Curve

The measurement system employs HATS (Head and Torso Simulator) compliant with ITU-T P.57 specifications and sound level meters meeting IEC 61672 standards to ensure objectivity and repeatability. Measurement microphones are positioned at driver, front passenger, and rear occupant locations with consistent horizontal alignment and elevation to minimize positional deviations.

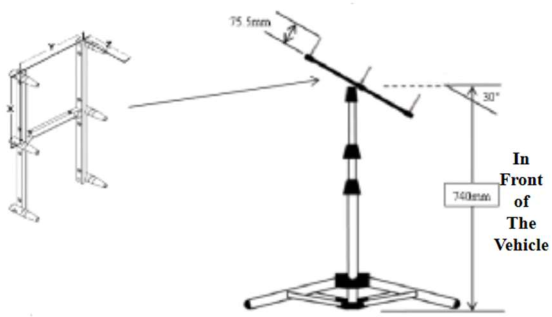

Figure 8. HATS Layout Diagram

4.2. Experimental Preparation

To ensure the comparability and reproducibility of results across various testing items, this study employed a microphone array and a Head and Torso Simulator (HATS) for measurements, with the experimental setup illustrated in Figures 8.

In different test scenarios, the microphone array and HATS were positioned at the driver's seat, front passenger seat, leftmost second-row seat, and rightmost second-row seat to cover typical vehicle seating positions and ensure measurement representativeness. Adjustable seats were set to their midpoints in both horizontal and vertical directions prior to testing and clearly marked; if seatbacks were adjustable, they were fixed in a vertical position. Once testing commenced, these seat positions remained unchanged. The front passenger seat configuration mirrored that of the driver's seat, and if rear seats were adjustable, they were handled identically to the driver's seat. The seat center point was determined according to NVH standards, and the bracket center point was aligned with the HATS center point to achieve consistent positioning reference. To eliminate systematic errors arising from geometric discrepancies, the height, horizontal position, and distance from the headrest of both the microphone array and HATS were rigorously maintained identical across all four seating positions.

During HATS-based testing, the center of the HATS simulated ears was aligned with the frontal center of the seat headrest, and the HATS was securely fastened using a seat belt to prevent movement during testing, thereby enhancing measurement stability and data reliability.

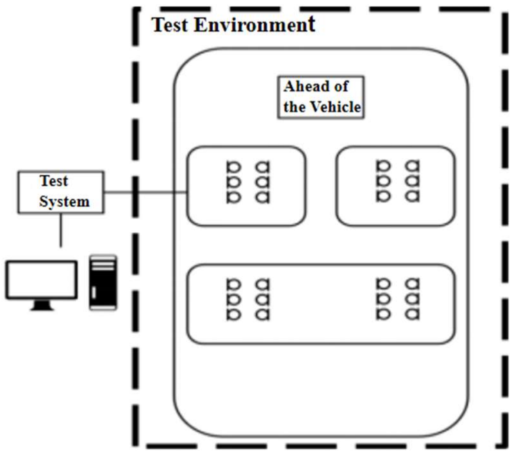

Figure 9. Measurement Microphone Array

4.3. Vehicle Preparation

Prior to testing, vehicle conditions were standardized to eliminate interference from non-test variables. Specific requirements include:

(1)Vehicle Load: Only test personnel and essential equipment are permitted; no additional load should be present.

(2)Windows and Door Locks: All must remain fully closed.

(3)Seat Configuration: Adjustable seats should be positioned at midpoints in both horizontal and vertical orientations, with seatbacks set as vertically as feasible.

(4)System Settings: Auxiliary components such as wipers and turn signals must be deactivated. Automated audio devices and other systems potentially interfering with acoustic measurements should remain inactive.

These standardized procedures ensure comparability of test results across different vehicles and operational conditions:a) Vehicle load: No additional load beyond test personnel and necessary equipment.b) Vehicle configuration: All doors and windows closed. Adjustable seats positioned at horizontal and vertical midpoints with maximally vertical seatbacks. Deactivation of wipers, turn signals, automated audio devices, and other systems affecting measurements.

4.4. System Latency Measurement

Delay measurement for in-vehicle audio systems employs Bluetooth transmission, with the testing procedure as follows:

(1) Signal connection and selection: The in-vehicle audio system is connected to the test system via Bluetooth, using the Composite Source Signal (CSS) defined by ITU-T P.501 as the test signal. The PN (pseudo-random noise) sequence length should exceed the maximum expected delay. A PN sequence of 16k samples (at 48 kHz sampling rate) or an equivalent signal is utilized.

(2) HATS calibration: The Head and Torso Simulator (HATS) undergoes free-field equalization according to ITU-T P.581, with its inner ear equalized output signal serving as the reference input.

(3) Delay calculation: The total system transmission delay is determined by computing the maximum cross-correlation function between the test signal and source signal. After subtracting the inherent delay of the test system itself, the actual delay of the in-vehicle audio system is obtained.

(4) Repeated measurements and averaging: Five repeated measurements are conducted per vehicle. After removing significant outliers, the arithmetic mean is taken as the final result.

4.5. Test Results and Analysis

Based on the testing and evaluation methods described in the previous chapter, this study selected touch interaction components from three vehicle models available on the market for testing and evaluation. The results indicate:

Distortion and sound quality scores are generally high, suggesting that current in-vehicle audio systems have achieved considerable maturity in hardware tuning and frequency response control. Some models demonstrate excellent harmonic distortion control and high acoustic reproduction fidelity, while others exhibit deficiencies in energy perception and spatial uniformity, negatively impacting overall sound quality. The adoption rates of active noise cancellation and independent sound zone functionalities remain low, with most models lacking corresponding modules, resulting in zero scores for these metrics. This reflects that active control in intelligent cockpit acoustics is still in its early stages of promotion. Furthermore, adaptive sweet spot functionality is only implemented in a few premium models, indicating that its productization remains constrained by algorithmic and computational limitations.

Table 1. Test Results

Vehicle Model	Distortion	Active Noise Cancellation Efficiency	Sound Quality	Independent Sound Zone	Adaptive Sweet Spot
1#	100	0	100.00	0	100
2#	75	0	94.50	0	0
3#	75	0	84.17	0	100

Comprehensive analysis reveals that the development of intelligent cockpit sound systems demonstrates a trend of "mature acoustic performance with intelligent functionalities yet to be enhanced." Current systems adequately meet fundamental auditory comfort and reproduction requirements but still require optimization in noise management, multi-zone audio playback, and personalized spatial perception. The testing and evaluation framework proposed in this study integrates subjective auditory perception with objective data through multidimensional weighting, effectively reflecting the comprehensive performance of intelligent sound systems and providing quantitative references for subsequent system calibration and technical improvements.

5. Conclusion

This study proposes a comprehensive testing and evaluation methodology centered on perceptual dimensions for in-vehicle intelligent cockpit sound systems, addressing user experience requirements while balancing acoustic performance and interactive features. Based on systematic functional characteristics, the research integrates acoustic signal processing with subjective perception assessment to establish a multidimensional evaluation framework encompassing distortion, sound quality, active noise cancellation, independent acoustic zones, and adaptive sweet spot localization. By incorporating user experience elements into a systematic analytical structure through weighted design, the method utilizes quantitative test results to derive comprehensive system scores via weighted calculations. This enables cross-

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model comparative analysis and performance optimization recommendations, providing scientific foundations for the design and calibration of automotive sound systems.

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