JOINT ADAPTIVE IMPULSE RESPONSE ESTIMATION AND INVERSE FILTERING FOR ENHANCING IN-CAR AUDIO

Ajay Dagar, Satyavolu Sai Nitish and Rajesh Hegde

Indian Institute of Technology Kanpur, India
Email: {adagar, rhegde}@iitk.ac.in

ABSTRACT

Performance of conventional audio equalization methods for improving in-car audio listening experience is limited by the uncertainties in computing the highly varying in-car channel response. Hence these methods generally compute the channel response which is then utilized in designing the inverse filter. In this paper, a novel adaptive equalization method is developed where the channel impulse response and inverse filter are jointly estimated. The method iteratively estimates the uncertainties in the channel response using a Kalman filter and updates the inverse filter gains at every step. The joint estimation method is thus adaptive and robust to the highly varying in-car acoustic conditions. Additional contributions of this work include the development of a car database that captures impulse responses and noise samples under various in-car conditions. Both subjective and objective evaluations are performed to show the performance improvements obtained using the proposed method.

Index Terms— adaptive equalization, car acoustics, inverse filtering, audio enhancement.

1. INTRODUCTION

In-car audio environment is considered to be one of the most preferred listening space globally [1]. With the change in technology and recent advancements in automobile infotainment systems, the in-car listening experience has been enhanced to a large extent but is limited by the unsteady acoustic conditions inside the car cabin. The presence of multiple reflective surfaces, interior cabin noise and complex acoustic channel behaviour accounts for the degradation in the quality of in-car audio [2]. In an attempt to minimize the in-car cabin noise, many passive and active noise cancelation strategies are proposed in literature [3]-[4]. In order to deal with reverberation and complex acoustic channels, an extensive research has also evolved in the field of audio equalization over the last two decades [5]-[6].

Many audio equalization strategies developed in literature are well suited for enhancing in-car audio listening experience [7]-[8]. In [7], a fixed equalization method based on channel inversion algorithms with desired temporal and spectral modifications is presented. In [8], a multipoint equalization obtained by the fractional octave smoothing of magnitude spectrum of car impulse response is developed. Other equalization techniques [9]-[11], based on inverse filtering can also be integrated within a car audio system. In [9], an exact multipoint equalization approach MINT (multiple-input/multiple-output inverse theorem) in room environment with limit on equalization point to be lesser than the number of loudspeakers is discussed. However in [10], solution to the classical crosstalk cancelation and equalization (CTCE) problem, consisting of single listener and two loudspeakers, is studied in the presence of multiple loudspeakers to provide exact equalization. The same work is extended to multiple listeners in [11]. So far, the above mentioned equalization methods provides single/multi-point audio equalization as required inside the car cabin but none of the approaches deals with the uncertainties in the channel responses. Utilization of these algorithms may not provide efficient audio equalization under time-varying channel conditions, as mentioned in [12]. In this paper, a joint adaptive impulse response estimation and inverse filtering method to enhance in-car audio is presented. In addition, a car database is developed to test the effectiveness of the proposed method. The in-car database is created by capturing the channel impulse responses and noise samples under various conditions of the car environment.

The rest of paper is organized as follows: In Section II, in-car system model for audio equalization is discussed. Joint adaptive impulse response estimation and inverse filtering method is discussed in Section III. Subjective and objective evaluations are presented in Section IV. Finally, Section V provides conclusion and future scope of the work.

2. IN-CAR AUDIO EQUALIZATION

In-car audio is affected by the dynamic channel behavior generating significant distortions in the signals being played through loudspeakers before it reaches the human ears. A system model that can be incorporated to overcome this limitation is discussed first. Subsequently, the inverse filtering method for audio equalization is discussed.

2.1. In-Car Audio System Model

Consider a car audio system that has two front speakers and a passenger with two ears acting as the sensors. The received signal at the sensor pair, $\mathbf{y}(n) = [y_l(n) \;\; y_r(n)]^T$, at discrete time index n, is given by

$$y_j(n) = \mathbf{x}_l(n) * \mathbf{h}_{lj}(n) + \mathbf{x}_r(n) * \mathbf{h}_{rj}(n), \quad \forall j \in \{l, r\} \qquad (1)$$

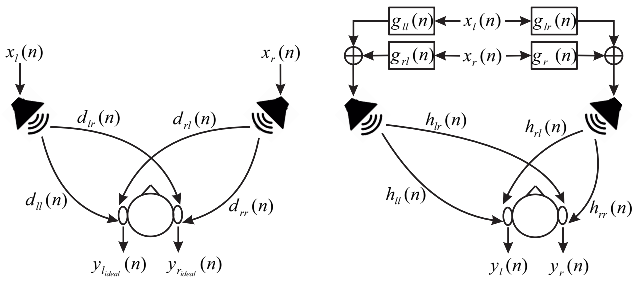
where, $\mathbf{x}_l(n)$ and $\mathbf{x}_r(n)$ are the input signals played at the left and right speakers, respectively. $\mathbf{h}_{ij}(n)$ is a time varying acoustic multipath channel of length L between i^{th} loudspeaker and j^{th} microphone and can be written as

$$\mathbf{h}_{ij}(n) = \mathbf{h}_{ij}(n-1) + \mathbf{u}_{ij}(n) \qquad (2)$$

where $\mathbf{u}_{ij}(n)$ is considered to be a zero-mean Gaussian noise vector with autocorrelation matrix as $R_{\mathbf{u}_{ij}}(n) = \sigma^2_{u_{ij}}(n)\mathbf{I}_L$.

The objective here is to iteratively estimate these time-varying channels, $\hat{\mathbf{h}}_{ij}(n)$, using an adaptive filter and obtain a flat-frequency channel response by implementing an inverse filter, whose gains are updated at every time index.

This work was funded by the SERB - DST under project no. SERB/EE/2017242



(a) (b)

Fig. 1: Figure illustrating in-car audio environment. (a) An ideal stereo system and (b) Equalization model for car stereo system.

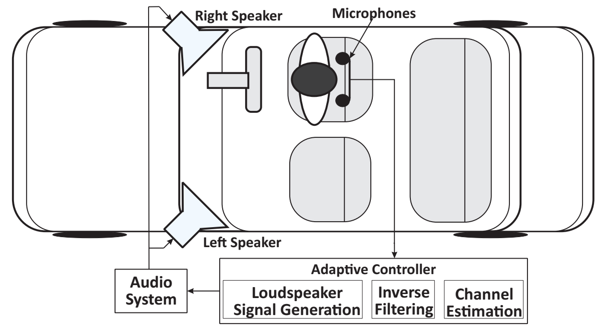


Fig. 2: Block diagram of the proposed in-car method.

2.2. Inverse Filtering For Audio Equalization

In general, a car stereo system differs from an ideal stereo system in many ways, as shown in Figure 1a. Firstly, it undergoes reverberation and has additive noise present at the sensors. Secondly, the listener has an asymmetrical sitting position with respect to the speakers resulting in an unequalized audio signals at the microphones. Therefore, the aim here is to equalize the car audio to match it as closely as possible with an ideal stereo system.

The equalization strategy for the stereo case is depicted in Figure 1b. Similar to [10], the input signals at each loudspeaker are preprocessed through the new FIR filters $\mathbf{g}_{ll}(n)$, $\mathbf{g}_{lr}(n)$, $\mathbf{g}_{rl}(n)$ and $\mathbf{g}_{rr}(n)$, each of length K, and can be expressed as

$$\mathbf{x'}_i(n) = \mathbf{x}_l(n) * \mathbf{g}_{li}(n) + \mathbf{x}_r(n) * \mathbf{g}_{ri}(n), \quad \forall i \in \{l, r\} \tag{3}$$

The recorded signals at the sensors can be re-written as

$$y_j(n) = \mathbf{x}_l(n) * \mathbf{h'}_{lj}(n) + \mathbf{x}_r(n) * \mathbf{h'}_{rj}(n), \quad \forall j \in \{l, r\} \tag{4}$$

where, $\mathbf{h'}_{ij}(n) = \mathbf{h}_{lj}(n) * \mathbf{g}_{il}(n) + \mathbf{h}_{rj}(n) * \mathbf{g}_{ir}(n), \quad \forall (i, j)$. In order to equalize the channel responses in (4) to desired channel responses, as in Figure 1a, it is required to have

$$\begin{bmatrix} \mathbf{h'}_{ll}(n) & \mathbf{h'}_{lr}(n) \\ \mathbf{h'}_{rl}(n) & \mathbf{h'}_{rr}(n) \end{bmatrix} = \begin{bmatrix} \mathbf{d}_{ll}(n) & \mathbf{d}_{lr}(n) \\ \mathbf{d}_{rl}(n) & \mathbf{d}_{rr}(n) \end{bmatrix} = \mathbf{D}(n) \tag{5}$$

Equivalently, we can write

$$\mathbf{H}(n)\mathbf{G}(n) = \mathbf{D}(n) \tag{6}$$

where, $\mathbf{H}(n) = \begin{bmatrix} \mathbf{H}_{ll}(n) & \mathbf{H}_{rl}(n) \\ \mathbf{H}_{lr}(n) & \mathbf{H}_{rr}(n) \end{bmatrix}, \mathbf{G}(n) = \begin{bmatrix} \mathbf{g}_{ll}(n) & \mathbf{g}_{rl}(n) \\ \mathbf{g}_{lr}(n) & \mathbf{g}_{rr}(n) \end{bmatrix}$. with $\mathbf{H}_{ij}(n)$ is a Sylvester matrix of size $(L+K-1) \times K$. In order to obtain inverse filters in (3), $2 \times 2(L+K-1)$ linear equations in (6) need to be solved for $2 \times 2K$ unknown variables. Since, $\mathbf{H}(n)$ is not square, as $2(L+K-1) > 2K$, an exact solution may not exist making it impossible to obtain the exact causal inverse filters. For this purpose, various approximation methods for finding $\mathbf{H}^{-1}(n)$ can be utilized, as discussed in the next sections.

3. JOINT ADAPTIVE IMPULSE RESPONSE ESTIMATION AND INVERSE FILTERING

In order to improve in-car audio under dynamic conditions, it is necessary to jointly perform adaptive impulse response estimation and apply inverse filtering at every time step. The block diagram of the proposed scheme is shown in Figure 2. In first step, channel response $\mathbf{h}_{ij}(n)$ are adaptively estimated by the use of Generalized Kalman Filter [13]. Later, the result is utilized to update the inverse filter gains, $\mathbf{g}_{ij}(n)$.

Let $\hat{\mathbf{h}}_j(n)$ represent the estimate of the channel state vector $\mathbf{h}_j(n) = \begin{bmatrix} \mathbf{h}_{lj}(n) & \mathbf{h}_{rj}(n) \end{bmatrix}^T$ at j-th sensor. As in [14], we can write

$$\hat{\mathbf{h}}_j(n) = \hat{\mathbf{h}}_j(n-1) + \mathbf{k}_j(n) e_j(n) \tag{7}$$

where, $\mathbf{k}_j(n)$ is the Kalman gain vector and $e_j(n)$ is the a priori error at n-th time instance given by

$$e_j(n) = y_j(n) - \mathbf{x}_f^T(n)\hat{\mathbf{h}}_j(n-1) \tag{8}$$

$$= \mathbf{x}_f^T(n)\mathbf{m}_j(n) + v_j(n) \tag{9}$$

Here, $\mathbf{m}_j(n) = \mathbf{h}_j(n) - \hat{\mathbf{h}}_j(n-1)$ is known as the a priori misalignment of the state vector $\mathbf{h}_j(n)$, $v_j(n)$ is the j-th sensor noise, $\mathbf{x}_f(n)$ is the modified input vector containing the L most recent samples each from $\mathbf{x'}_l(n)$ and $\mathbf{x'}_r(n)$ as defined in (3). The Kalman filter estimates $\hat{\mathbf{h}}_j(n)$ by minimizing the cost function given by

$$J(n) = \frac{1}{2L} tr[\mathbf{R}_{\mu_j}(n)] \tag{10}$$

where $\mathbf{R}_{\mu_j}(n)$ is the autocorrelation matrix of the posterior misalignment of $\mathbf{h}_j(n)$, i.e. $\boldsymbol{\mu}_j(n) = \mathbf{h}_j(n) - \hat{\mathbf{h}}_j(n)$, and can be obtained from the relation between $\boldsymbol{\mu}_j(n)$ and $\boldsymbol{m}_j(n)$, given by

$$\boldsymbol{m}_j(n) = \boldsymbol{\mu}_j(n) + \mathbf{u}_j(n) \tag{11}$$

where, $\mathbf{u}_j(n)$ represent uncertainties in $\mathbf{h}_j(n)$, corresponding to equation (2), with autocorrelation as a diagonal matrix, $\mathbf{R}_{\mathbf{u}_j}(n) = diag\{\mathbf{R}_{\mathbf{u}_{lj}}(n), \mathbf{R}_{\mathbf{u}_{rj}}(n)\}$. Solving the optimization problem, in (10), results in Kalman filter with the following update equations

$$\mathbf{R}_{\mathbf{m}_j}(n) = \mathbf{R}_{\mu_j}(n-1) + \mathbf{R}_{\mathbf{u}_j}(n) \tag{12}$$

$$\sigma^2_{e_j}(n) = \mathbf{x}_f^T(n)\mathbf{R}_{\mathbf{m}_j}(n)\mathbf{x}_f(n) + \sigma^2_{v_j} \tag{13}$$

$$\mathbf{k}_j(n) = \frac{1}{\sigma^2_{e_j}(n)} \mathbf{R}_{\mathbf{m}_j}(n)\mathbf{x}_f(n) \tag{14}$$

$$e_j(n) = y_j(n) - \mathbf{x}_f^T(n)\hat{\mathbf{h}}_j(n-1) \tag{15}$$

$$\hat{\mathbf{h}}_j(n) = \hat{\mathbf{h}}_j(n-1) + \mathbf{k}_j(n)e_j(n) \tag{16}$$

$$\mathbf{R}_{\mu_j}(n) = [\mathbf{I}_{2L} - \mathbf{k}_j(n)\mathbf{x}_f^T(n)]\mathbf{R}_{\mathbf{m}_j}(n) \tag{17}$$

After estimating the channel impulse responses $\hat{\mathbf{h}}_j(n)$, the inverse filter gains $\hat{\mathbf{g}}_j(n)$ can be estimated by solving equation (6). Least square method is the best estimator that can provide a close and an approximate solution but it involves finding $\mathbf{H}^{\dagger}(n)$, pseudo inverse matrix, which is computationally costly to perform repeatedly. Therefore, it is necessary to use methods that are simpler in terms of computational efficiency. One such algorithms uses the steepest descent method to estimate the inverse filter [15].

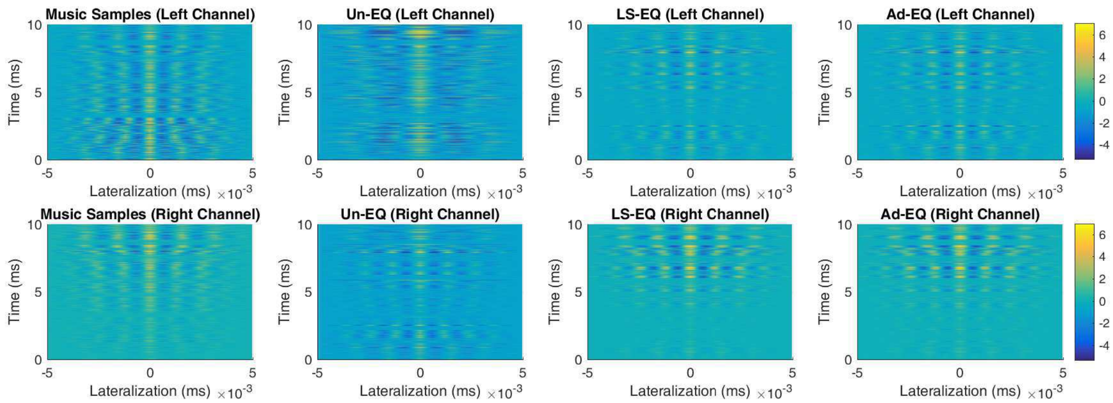


Fig. 3: Figure illustrating the correlograms of sound samples at SNR = 10dB and RPM = 1000. (a) Original stereo sound, (b) Un-equalized, (c) LS-equalized and (d) Ad-equalized.

Fig. 4: Figure illustrating the experimental setup with dummy placed at drivers' location with microphones placed inside bionic ears.

RPM	Method	PEAQ	DI	PSM	PSMt
1000	Un-EQ	-3.7392	-2.7989	0.8238	0.3215
	LS-EQ	-3.5126	-2.0777	0.8695	0.5987
	Ad-EQ	-3.5441	-2.1559	0.8612	0.5979
3000	Un-EQ	-3.8939	-3.8560	0.8002	0.2974
	LS-EQ	-3.7629	-2.9094	0.8598	0.5786
	Ad-EQ	-3.7991	-3.1009	0.8585	0.5760

Table 1: PEAQ and PSMt scores of equalization audio in simulated environment

3.1. Computing Inverse Filter Gains using Steepest Descent

Considering $\mathbf{H}(n)$ and the inverse filter estimate $\hat{\mathbf{G}}(n)$ in (6) at discrete time n, the aim is to minimize the cost function

$$\begin{aligned}\mathbf{J}(n) &= \|\mathbf{H}(n)\hat{\mathbf{G}}(n) - \mathbf{D}(n)\|_F^2 \\ &= \|\mathbf{H}(n)\hat{\mathbf{g}}_l(n) - \mathbf{d}_l(n)\|_2^2 + \|\mathbf{H}(n)\hat{\mathbf{g}}_r(n) - \mathbf{d}_r(n)\|_2^2\end{aligned} \tag{18}$$

where, $\hat{\mathbf{g}}_i(n)$ and $\mathbf{d}_i(n)$, for $i \in \{l, r\}$, are the i^{th} column of $\hat{\mathbf{G}}(n)$ and $\mathbf{D}(n)$, respectively. Therefore, minimization of cost function $\mathbf{J}(n)$, is equivalent to minimize each term in (18). Hence, the inverse filter can be adaptively estimated by

$$\hat{\mathbf{g}}_i(n+1) = \hat{\mathbf{g}}_i(n) - 2\mu\mathbf{H}^T(n)(\mathbf{H}(n)\hat{\mathbf{g}}_i(n) - \mathbf{d}_i(n)) \tag{19}$$

The above update rule can be used to efficiently adapt the estimated inverse filter, $\hat{\mathbf{G}}(n) = [\hat{\mathbf{g}}_l(n) \;\; \hat{\mathbf{g}}_r(n)]$, at every step. By estimating the system impulse response iteratively and re-estimating the inverse filter at each discrete time index, an adaptive equalization can be achieved.

4. PERFORMANCE EVALUATION

In this section, the experimental setup and data acquisition methodology used for developing the in-car audio database are described first. Subsequently, the experimental conditions for performance evaluations of the proposed method are discussed. Finally, the subjective and objective evaluations are presented to illustrate the performance of the proposed method.

4.1. Development of In-Car Audio Database

The in-car audio database consists of two components. First, it comprises of the measurements of the channel impulse responses at specific location inside the car cabin. Second, it consists of noise samples recorded at different engine rotations per minute (RPMs).

4.1.1. In Car Audio Setup

For recording the in-car database, the acoustic condition inside Tata Indigo car is considered. While measuring the channel impulse responses, the car is parked at a silent/noise free location to minimize the distortions in the recorded signals. Figure 4, shows the in-car audio setup used for developing the database. The car audio system consist of stereo speakers placed at the front-right and front-left corners. For recording purpose, a dummy with two microphones, inserted in its ears, is placed at the driver position. The engine of the car is turned off while recording the impulse responses and turned on when the music signals and the noise samples are measured.

4.1.2. In Car Audio Data Acquisition

The impulse response of any acoustic scene can be measured by various methods present in literature [16] - [17]. In this work, maximum length sequence (MLS) method for measuring the acoustic channel is used. For the same, a software IR Measurement Tool is developed in MATLAB using Graphical User Interface Design Environment (GUIDE) framework that incorporates the MLS method and provide a pseudo-random sequence of user specific order and repetitions. The sequence is then used to excite in-car loudspeakers. The output at the desired microphone is recorded and then deconvolved to get an estimate of the desired channel impulse response. For measuring the noise samples, the car engine is made to run at 1000 rpm, 1500 rpm, 2000 rpm, 2500 rpm and 3000 rpm and corresponding responses are recorded with sampling rate of $Fs = 44.1$KHz.

4.2. Experimental Conditions

The performance of the proposed method is evaluated in a simulated environment where a music signal is convolved with the measured impulse responses to obtain the observed signal corresponding to the

RPM	Methods	Loudspeaker								Headphone							
		Bass		Treble		Pleasantness		Perception of Motion		Bass		Treble		Pleasantness		Perception of Motion	
		μ	σ^2	μ	σ^2	μ	σ^2	μ	σ^2	μ	σ^2	μ	σ^2	μ	σ^2	μ	σ^2
1000	Un-EQ	3.4	0.79	2.4	0.43	3.1	0.77	2.1	0.71	3.0	1.05	2.6	0.83	3.3	0.54	2.4	0.61
	LS-EQ	3.6	0.63	3.3	0.71	4.2	0.54	2.5	0.38	4.0	0.81	3.6	1.26	4.4	1.19	3.2	0.81
	Ad-EQ	3.6	0.63	3.1	0.90	4.0	0.71	2.4	0.59	3.8	0.97	3.3	0.65	4.1	0.76	3.1	0.89
3000	Un-EQ	2.1	1.05	2.1	0.58	2.4	0.71	1.8	0.90	2.4	0.96	2.2	1.18	2.1	0.63	2.1	1.21
	LS-EQ	2.8	1.19	3.1	0.87	3.3	0.64	2.4	0.95	3.1	0.82	3.4	0.70	3.7	0.49	3.2	0.81
	Ad-EQ	2.8	1.19	2.9	0.63	3.1	0.53	2.2	0.72	3.0	0.68	3.2	0.43	3.4	0.75	3.1	0.89

Table 2: Mean Opinion Scores (MOS) obtained for sound samples evaluated at SNR = 10dB and different engine RPMs

Algorithm 1 Adaptive equalization algorithm for enhancing the quality of in-car audio

Initialize $\hat{\mathbf{h}}_j(0)$ using MLS method
Initialize $\hat{\mathbf{g}}_j(0)$ to the least squares inverse of $\hat{\mathbf{h}}_j(0)$
Initialize constants $\sigma^2_{v_j}$ and $\sigma^2_{u_j}$
$n \leftarrow 0$
while terminate == false **do**
  Compute the filtered input signal $\mathbf{x}_f(n)$ using (3)
  Estimate $\mathbf{R}_{\mathbf{m}_j}(n)$ using (12)
  Estimate $\sigma^2_{e_j}(n)$ using (13)
  Compute the Kalman gain vector $\mathbf{k}_j(n)$ using (14)
  Compute the error signal $e_j(n)$ using (15)
  Adapt the impulse response estimate $\hat{\mathbf{h}}_j(n)$ using (16)
  Compute $\mathbf{R}_{\mu_j}(n)$ using (17)
  Adapt the inverse filter estimate $\hat{\mathbf{g}}_j(n)$ using (19)
end while

microphone location, as mentioned in section 4.1.1. The channel responses $\mathbf{h}_{ij}(n)$ are varied according to equation (2). Initially the Kalman filter is adapted by the measured channel response and corresponding inverse filter is applied to the music file before playing. Later to capture the uncertainties arising in the channel impulse responses, the Kalman filter tries to estimate $\hat{\mathbf{h}}_{ij}(n)$ using algorithm-1. As mentioned in [13], $\sigma^2_{u_j}(n)$ plays an important role between good tracking and low misalignment during channel estimation. Therefore, $\sigma^2_{u_j}(n) = 10^{-12}$ was chosen for this experiment. $\sigma^2_{v_j}(n)$ can be easily estimated from the noise samples recorded in the car database. The experiment is repeated for the various engine RPMs to evaluate the performance of overall system design.

4.3. Objective Evaluations

The perceived audio quality is evaluated by measuring the PEAQ and PSMt scores [18]-[19] that compares the original sound track with the simulated unequalized, Un-EQ, and adaptively equalized, Ad-EQ and LS-EQ, as showm in Table-1. From Table-1 it can be seen that, with increasing RPMs the overall difference grade (ODG) value under PEAQ evaluation shows improved performance for both LS-EQ and Ad-EQ methods. The distortion index (DI) values, PSM and PSMt values reflects the similar trend indicating effective improvement of equalized sounds over unequalized soundtracks.

In Figure 3, correlation based objective evaluation is also presented in support of proposed approach. Each channel of the original sound and simulated sounds are autocorrelated over time-frames with lateralization of 5 milliseconds. From the Figure 3, it is visible that the correlogram of the unequalized sound samples is scattered and dispersed along the lag values whereas adaptively equalized sound sample have similar correlogram to the original samples.

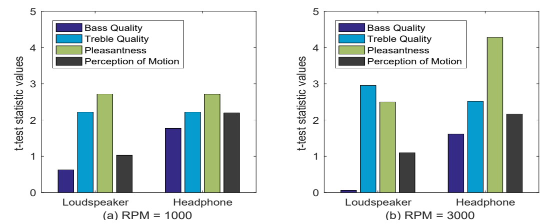


Fig. 5: Bar plots corresponding to Welch's t-test for different test setups at various engine RPMs and SNR = 10dB.

4.4. Subjective Evaluations

The described adaptive in-car method is also evaluated subjectively by 10 human subjects on the basis of Mean Opinion Scores (MOS). Each subject is made to listen the simulated unequalized and adaptive equalized stereo sound for two test scenarios, i.e. loudspeaker setup and headphone setup, and were asked to rate the audio quality on a standard scale of 1 to 5 for different evaluations parameters [20] like quality in terms of bass retention, treble retention, pleasantness and perception of motion.

The resulting scores are then averaged across different subjects to obtain the MOS scores as presented in Table-2. From the MOS values, it can be observed that perception of motion and the high frequency component (timbre) of the original stereo sound are mostly effected by the time-varying car impulse responses. An improvement in these parameters can be seen when the system is adapted with the proposed method. For analyzing subjective scores, 2-tail paired Welch's t-test was conducted comparing the adaptively-equalized sound with unequalized sound for both test setups and at different RPMs, as shown in Figure 5. At low RPMs in both the test setups, the null hypothesis is rejected for pleasantness while t-values for treble are obtained at the boundaries of confidence interval. T-values for perception of motion shows improvement when subjects are made to listen through headphones.

5. CONCLUSION

In this work, a method of adaptive in-car audio equalization by jointly estimating the channel impulse response and the inverse filter is presented. The method is robust to the variations in the characteristic of the car audio system and can work online unlike the other conventional methods. Both the subjective and objective evaluations indicates the improved performance. In addition, an in-car audio database has been developed that can be used to study the in-car acoustic characteristics in simulated environment. The adaptive method presented here is limited to two sensors placed near the listeners' ears which can be extended to multiple sensors for exact equalization by finding the optimal microphone locations inside a car cabin. Further, the proposed work can also be extended to multiple listening sweet spots inside a car.

6. REFERENCES

[1] Floyd E Toole, *Sound reproduction: Loudspeakers and rooms*, Taylor & Francis, 2008.

[2] Roger Shively, "Automotive audio design (a tutorial)," in *Audio Engineering Society Convention 109*. Audio Engineering Society, 2000.

[3] Sen M Kuo and Dennis R Morgan, "Active noise control: a tutorial review," *Proceedings of the IEEE*, vol. 87, no. 6, pp. 943–973, 1999.

[4] Hanchi Chen, Prasanga Samarasinghe, and Thushara D Abhayapala, "In-car noise field analysis and multi-zone noise cancellation quality estimation," in *Signal and Information Processing Association Annual Summit and Conference (APSIPA), 2015 Asia-Pacific*. IEEE, 2015, pp. 773–778.

[5] Darren B Ward, "On the performance of acoustic crosstalk cancellation in a reverberant environment," *The Journal of the Acoustical Society of America*, vol. 110, no. 2, pp. 1195–1198, 2001.

[6] Takafumi Hikichi, Marc Delcroix, and Masato Miyoshi, "Inverse filtering for speech dereverberation less sensitive to noise and room transfer function fluctuations," *EURASIP Journal on Applied Signal Processing*, vol. 2007, no. 1, pp. 62–62, 2007.

[7] Angelo Farina and Emanuele Ugolotti, "Spatial equalization of sound systems in cars," in *Audio Engineering Society Conference: 15th International Conference: Audio, Acoustics & Small Spaces*. Audio Engineering Society, 1998.

[8] S Cecchi, L Palestini, P Peretti, F Piazza, and A Carini, "Multipoint equalization of digital car audio systems," in *Image and Signal Processing and Analysis, 2009. ISPA 2009. Proceedings of 6th International Symposium on*. IEEE, 2009, pp. 650–655.

[9] Masato Miyoshi and Yutaka Kaneda, "Inverse filtering of room acoustics," *IEEE Transactions on acoustics, speech, and signal processing*, vol. 36, no. 2, pp. 145–152, 1988.

[10] Yiteng Huang, Jacob Benesty, and Jingdong Chen, "On crosstalk cancellation and equalization with multiple loudspeakers for 3-d sound reproduction," *IEEE Signal Processing Letters*, vol. 14, no. 10, pp. 649–652, 2007.

[11] Yiteng Huang, Jacob Benesty, and Jingdong Chen, "Generalized crosstalk cancellation and equalization using multiple loudspeakers for 3d sound reproduction at the ears of multiple listeners," in *Acoustics, Speech and Signal Processing, 2008. ICASSP 2008. IEEE International Conference on*. IEEE, 2008, pp. 405–408.

[12] Biljana D Radlovic, Robert C Williamson, and Rodney A Kennedy, "Equalization in an acoustic reverberant environment: Robustness results," *IEEE Transactions on Speech and Audio Processing*, vol. 8, no. 3, pp. 311–319, 2000.

[13] Constantin Paleologu, Jacob Benesty, and Silviu Ciochină, "Study of the general kalman filter for echo cancellation," *IEEE Transactions on Audio, Speech, and Language Processing*, vol. 21, no. 8, pp. 1539–1549, 2013.

[14] Steven M Kay, "Fundamentals of statistical signal processing, volume i: estimation theory," 1993.

[15] Wancheng Zhang, Andy WH Khong, and Patrick A Naylor, "Adaptive inverse filtering of room acoustics," in *Signals, Systems and Computers, 2008 42nd Asilomar Conference on*. IEEE, 2008, pp. 788–792.

[16] Guy-Bart Stan, Jean-Jacques Embrechts, and Dominique Archambeau, "Comparison of different impulse response measurement techniques," *Journal of the Audio Engineering Society*, vol. 50, no. 4, pp. 249–262, 2002.

[17] Angelo Farina, "Advancements in impulse response measurements by sine sweeps," in *Audio Engineering Society Convention 122*. Audio Engineering Society, 2007.

[18] Thilo Thiede, William C Treurniet, Roland Bitto, Christian Schmidmer, Thomas Sporer, John G Beerends, and Catherine Colomes, "Peaq-the itu standard for objective measurement of perceived audio quality," *Journal of the Audio Engineering Society*, vol. 48, no. 1/2, pp. 3–29, 2000.

[19] Rainer Huber and Birger Kollmeier, "Pemo-qa new method for objective audio quality assessment using a model of auditory perception," *IEEE Transactions on audio, speech, and language processing*, vol. 14, no. 6, pp. 1902–1911, 2006.

[20] Angelo Farina and Emanuele Ugolotti, "Subjective evaluation of the sound quality in cars by the auralisation tehcnique," *PROCEEDINGS-INSTITUTE OF ACOUSTICS*, vol. 19, pp. 105–114, 1997.