Audio Engineering Society

Convention Paper 5844

Presented at the 114th Convention
2003 March 22–25 Amsterdam, The Netherlands

Comparison of Modal Equalizer Design Methods

Poju Antsalo[1], Matti Karjalainen[1], Aki Mäkivirta[2] and Vesa Välimäki[1]

[1]Helsinki University of Technology, Lab. of Acoustics and Audio Signal Processing, P.O. Box 3000, FIN-02015 HUT, Finland

[2]Genelec Oy, Olvitie 5, FIN-74100 Iisalmi, Finland

ABSTRACT

Modal equalization of low-frequency room modes has recently been proposed as a method to improve sound reproduction in spaces where modal decay time is too long. Modal equalization is achieved by signal processing reducing the pole radii of problematic modes in the overall transfer function. In this paper, we compare the performance of two proposed methods for designing modal equalizers. Comparison includes a preliminary subjective listening test indicating a possible marginal improvement by modal equalization over conventional magnitude equalization.

1. INTRODUCTION

A loudspeaker installed in a room acts as a coupled system where the room properties typically dominate the rate of energy decay. The characteristics of this energy decay are well known.

Passive methods of controlling the decay rate and properties of this energy decay are also straightforward and well established. At low frequencies these methods are unpractical or very expensive to implement because the physical size and cost of necessary absorbers increases rapidly with decreasing frequency.

Active control becomes feasible as the wavelengths become long and the sound field develops less diffuse [1-5]. Methods to optimize the response at a listening position by selecting suitable loudspeaker locations have been proposed [6] but cannot fully solve the problem. Because of these reasons, there has been an increasing interest in active methods of sound field control at low frequencies.

Modal resonances in a room can become audible as timbre changes because they modify the magnitude response of the primary sound or, when the primary sound ends, because they are no longer masked [7,8]. The ability to detect a modal resonance appears to be very dependent on the signal content. Olive et al. report that for mid and high frequencies low-Q resonances are more readily audible with continuous signals containing a broad frequency spectrum while high-Q resonances become more audible with transient discontinuous signals [8]. They also report detection thresholds for resonances for both continuous broadband sound and transient discontinuous sound.

At low Q values, antiresonances (notches) are as audible as resonances. Audibility of antiresonances reduces dramatically for wideband continuous signals as the Q value becomes high [8]. Detectability of resonances reduces by approximately 3 dB for each doubling of the Q value [7,8] and low Q resonances are more readily heard with zero or minimal time delay relative to the direct sound [7]. Duration of the reverberant decay in itself appears an unreliable indicator of the audibility of the resonance [7] as audibility seems to be more determined by frequency domain characteristics of the resonance.

We have previously proposed two methods for the estimation of modal parameters and using the estimated mode information to design filters that reduce the modal decay time [9-12]. In this paper we present a series of experiments where we have studied the applicability and performance of the proposed modal equalization techniques with case studies and measured listening room response data.

2. MODAL EQUALIZER DESIGN

We define modal equalization as a process that modifies the rate of modal decay. A modal resonance is represented in the z-domain transfer function as a pole pair with pole radius r and pole angle θ

$$H_{\mathrm{m}}(z) = \frac{1}{(1 - re^{j\theta} z^{-1})(1 - re^{-j\theta} z^{-1})} \qquad (1)$$

To shorten the decay time the Q value of the resonance must be decreased by moving the pole pair closer to the origin. Each mode is modeled by an exponential decay function

$$h_{\mathrm{m}}(t) = A_{\mathrm{m}} e^{-\tau_{\mathrm{m}} t} \sin(\omega_{\mathrm{m}} t + \phi_{\mathrm{m}}) \qquad (2)$$

where A_{m} is the initial envelope amplitude of the decaying sinusoid, τ_{m} is a coefficient that denotes the decay rate, ω_{m} is the angular frequency of the mode, and ϕ_{m} is the initial phase of the oscillation.

One method of implementing modal equalization is to modify the audio signal fed into a loudspeaker (Fig. 1). The total transfer function from the sound radiator to the listening position represented in z-domain is

$$H(z) = G(z) H_{\mathrm{m}}(z) \qquad (3)$$

where $G(z)$ is the transfer function of the sound radiator from the electrical input to acoustical output and $H_{\mathrm{m}}(z)$ is the transfer function of the path from the sound radiator to the listening position.

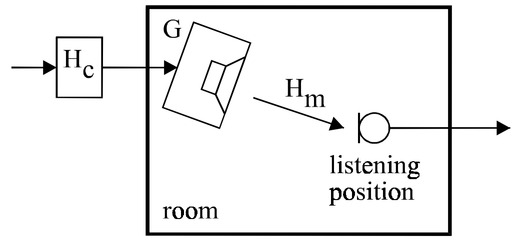

Fig. 1. A simple set-up for modal equalization.

An impulse response of the acoustic path measured from the loudspeaker to the listening position is needed in order to design a modal equalization filter. Modal equalization concentrates to frequencies below 200 Hz and within a restricted listening area. The process of modal equalization starts with the estimation of octave band reverberation times between 500 Hz and 2 kHz to calculate a mean reverberation time at mid frequencies, which is used as the basis for determining the target for maximum allowed low-frequency reverberation time. We define the target decay time relative to the mean decay time T_{60} in mid-frequencies, increasing for example by 0.2 s as the frequency decreases from 300 Hz down to 50 Hz.

We have previously developed two methods to design modal equalizer filters [9-11]. The first method (later called AMK) attempts to directly identify mode frequencies in the magnitude response and then to obtain the decay rate τ_{m} and mode frequency ω_{m} by using time-frequency presentation of the impulse response at frequencies below 200 Hz [9-12]. The decay rate for each identified mode frequency is calculated using a nonlinear fitting technique modeling the data as a sum of an exponential decay and background noise. Mode frequencies are estimated directly in the magnitude spectrum using a special detection function [12] for those frequency bins that show slow decay. The modal equalizer filter is then designed using the mode parameter data. The longest decay time is corrected by filtering with an equalizer, and the process is continued iteratively until all decay rates are within the desired bounds.

The second method (later called ARMA) [10,11] identifies the pole and zero pairs describing a modal resonance by fitting a pole-zero least-squares model directly to the room impulse response. Similarly to the AMK method, the longest decay rate detected by finding the pole closest to the unit circle is compensated by designing an equalizer filter for it, and the method is then iteratively applied until all decay rates are within the desired bound. The second method does not require the intermediate stage of determining values for the decay rate of a mode, and is better able to model closely spaced mode resonances.

2.1. A simple case with synthetic modes

The case study presented here clarifies the idea of modal equalization. Figure 2 shows the waterfall plot of a loudspeaker response measured in an anechoic chamber, with five synthetic modes added at frequencies 50, 55, 100, 130, and 180 Hz. The corresponding decay times are 1.4, 0.8, 1.0, 0.8, and 0.7 s. Now, we set up a modal equalizer design target to reduce these decay times to 0.30, 0.30, 0.26, 0.24, and 0.20 s. After processing the synthetic response with the equalizer designed by the AMK algorithm, the decay times have been reduced (Fig. 3). However, the decay at 50 and 55 Hz continues with the original rate after an initial rapid decay of 15-20 dB. This shows that the method is able to control the decay rate of individual modes, but its performance is limited when modes are close to each other in frequency.

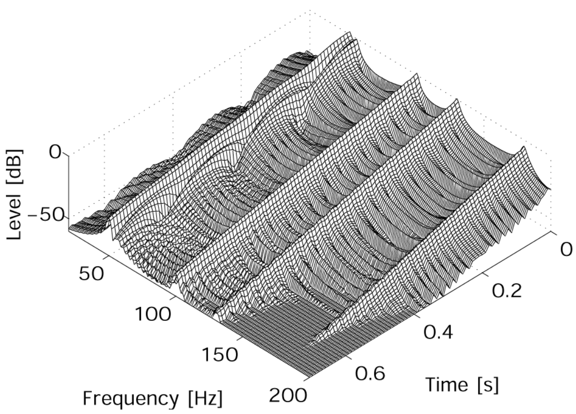

Fig. 2. Waterfall plot of five synthetic modes.

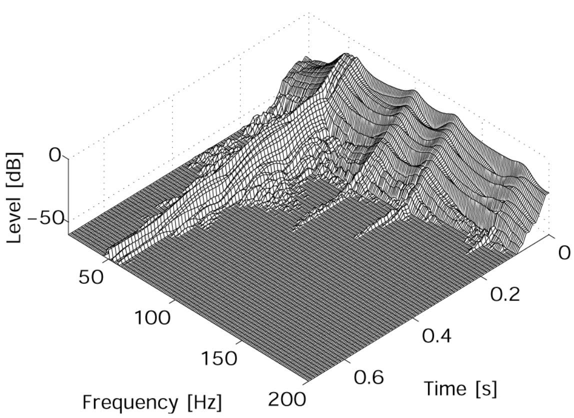

Fig. 3. Waterfall plot of five synthetic modes after modal equalization.

3. OBJECTIVE EVALUATION

The performance of AMK and ARMA equalizer design algorithms was studied with objective evaluation. Synthetic cases were created to observe how accurately the algorithms estimate the mode frequency and decay time values.

3.1. Synthetic responses

There are three different cases each representing an increasing “degree of difficulty” in the sense of how close the modal frequencies f are to each other (Tables 1-3). The decay time t for each mode is the same.

Table 1. Modal frequencies and decay times for synthetic response 1.

	f (Hz)	t (s)
Mode 1	50.4	0.6
Mode 2	100.4	0.6
Mode 3	140.4	0.6

Table 2. Modal frequencies and decay times for synthetic response 2.

	f (Hz)	t (s)
Mode 1	90.4	0.6
Mode 2	100.4	0.6
Mode 3	116.4	0.6

Table 3. Modal frequencies and decay times for synthetic response 3.

	f (Hz)	t (s)
Mode 1	99.4	0.6
Mode 2	100.4	0.6
Mode 3	102.4	0.6

3.2. Mode parameter estimation errors

The estimation error of mode frequency Δf and the error of the decay rate Δt are calculated using Eqs. 4 and 5,

$$\Delta f = \frac{|\, f - f' \,|}{f} \tag{4}$$

$$\Delta t = \frac{|\, t - t' \,|}{t} \tag{5}$$

where the actual mode frequency is denoted by f and decay time by t, and the estimated mode frequency by f' and decay time by t'.

When the modes are sparsely spaced in frequency, both equalizer design methods show similar performance in estimating parameters.

Estimation errors (Tables 4-6) tend to increase as modes move closer to each other, especially in the case of decay time estimated by the AMK design method. For closely spaced modes (response 3), the ARMA design method performs well, significantly better than the AMK method. This is due to the decay envelope fluctuation (beating) in level estimates of closely spaced modes producing energy-time curves more complex than simple exponential decays, causing an error when fitting data assuming an exponential decay model as in the AMK method.

With true room responses the performance of both design methods is limited mainly by the complexity of the modal characteristics, measurement anomalies and noise.

4. SUBJECTIVE LISTENING TEST

The perceptual benefit of modal equalization was studied with listening tests. The aim of the listening test was to demonstrate the incremental improvements provided by conventional magnitude equalization and modal equalization.

The responses of two representative rooms were measured and equalizers were designed for each room. Audio material with energy content at low frequencies were chosen and convolved with the equalizer filters to produce the audio test items.

Table 4. Synthetic case 1, estimation error percentages for the mode frequency and decay time.

	Equalizer design method			
	AMK		ARMA	
Mode	Δf	Δt	Δf	Δt
1	0.16%	2.28%	0.07%	3.8%
2	0.0082%	0.32%	0.005%	0.58%
3	0.017%	2.9%	0.0034%	0.60%

Table 5. Synthetic case 2, estimation error percentages for the mode frequency and decay time.

	Equalizer design method			
	AMK		ARMA	
Mode	Δf	Δt	Δf	Δt
1	0.19%	8.2%	0.015%	0.80%
2	0.17%	16%	0.008%	0.16%
3	0.075%	5.7%	0.001%	0.096%

Table 6. Synthetic case 3, estimation errors percentages for the mode frequency and decay time.

	Equalizer design method			
	AMK		ARMA	
Mode	Δf	Δt	Δf	Δt
1	0.53%	19%	0.057%	0.077%
2	0.12%	86%	0.036%	3.2%
3	0.27%	15%	0.020%	0.55%

Test subjects listened to audio test items over headphones. The reference was the sound sample convolved with the original room response. This was compared to the sound sample processed with a magnitude equalizer, or a combination of a modal equalizer design either by the AMK method or the ARMA method and a magnitude equalizer.

Subjects were asked to grade on a continuous scale two properties of the sound samples, the *temporal definition* and the *spectral balance*.

4.1. Room response measurements

To facilitate the subjective listening tests impulse responses of two real rooms were measured using an omnidirectional microphone and both ears of a dummy head[1]. The omnidirectional microphone measurement was used later in the design of the equalizer filters. The dummy head equalized impulse responses were convolved with audio test samples to produce subjective test items for headphone listening.

The two rooms are similar in size to typical professional audio monitoring spaces, and are referred to later as Room G and Room S. The response magnitude and waterfall plots of these rooms show distinct room modes (Fig. 4). The volume of room G is about 102 m^3 and room S about 90 m^3, and mid-frequency RT_{60} values for these rooms are 0.40 s and 0.77 s, respectively.

4.2. Design of room equalizers

Three room equalizers were designed for each room (Table 7). All equalizers were designed for the target frequency band of 20 Hz … 200 Hz using omnidirectional microphone measurements of the rooms.

[1] Manufacturer Cortex Electronics GmbH, model Manikin Mk2/NCF1.

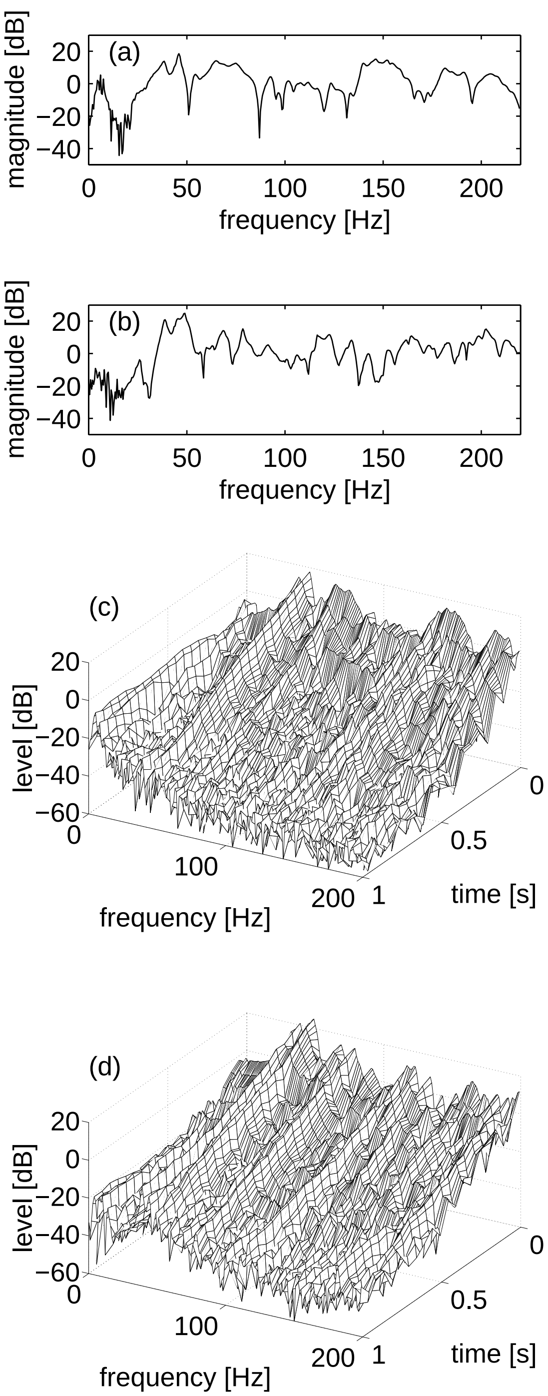

Fig. 4. Responses of rooms used in the subjective listening test, from top to bottom, (a) magnitude responses of Room G and (b) Room S, (c) waterfall responses of Room G and (d) Room S.

As a reference to current state-of-the-art, a magnitude equalizer was designed as the inverse of the third-octave smoothed magnitude response within the target frequency band. Such an equalizer ignores individual modes, but corrects the magnitude response unbalance within the target frequency range. This magnitude equalizer is later referred to with an acronym 'Ampleq' (Table 7).

The modal equalizers affect primarily the decay times, but do not necessarily provide magnitude equalization similar to Ampleq. Because of this, and because it has been recognized that broadband equalization is necessary after modal equalization [9,10], third-octave magnitude equalization was applied to both the AMK and the ARMA modal equalized systems. In this way the subjective listening test detects the incremental improvement of using modal equalization before magnitude equalization.

Equalizer magnitude responses of all three equalizer designs for rooms G and S are presented in Figures 9-11. Tables of poles and zeros (sampling frequency 11025 Hz) are given for each modal equalizer design (excluding the magnitude equalizers) in Tables 14-17 in appendix 1. Figures 5-8 show these poles and zeros visually.

Table 7. Descriptions of equalizers used in the subjective listening test.

Equalizer	Description
Ampleq	FIR inverse filter of the third-octave smoothed response within 20…200Hz
AMK	Type AMK modal equalizer followed by type 'Ampleq' magnitude equalizer designed on the AMK filtered data
ARMA	Type ARMA modal equalizer followed by type 'Ampleq' magnitude equalizer designed on the ARMA filtered data

AMK design method attempts to assign multiple notches to equalize certain types of modes, such as closely spaced modes and modes with non-exponential decay. Also, if the frequency or decay time of a mode is erroneously estimated, the AMK method compensate for this failure by assigning multiple equalizer notches close to such mode. The ARMA design method seems to be more successful in modeling and correcting these problematic mode structures.

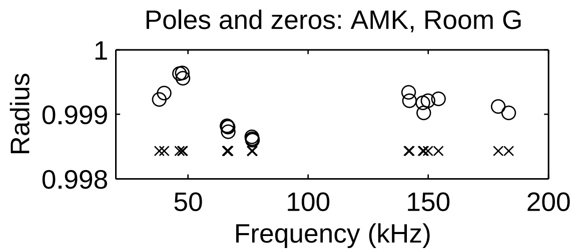

Fig. 5. Poles (×) and zeros (o) for the modal equalizer of AMK (room G).

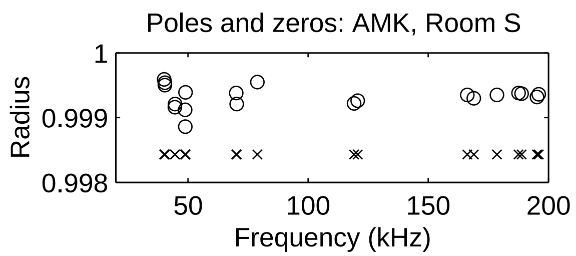

Fig. 6. Poles (×) and zeros (o) for the modal equalizer of AMK (room S).

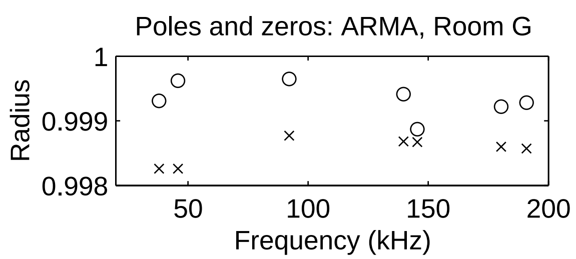

Fig. 7. Poles (×) and zeros (o) for the modal equalizer of ARMA (room G).

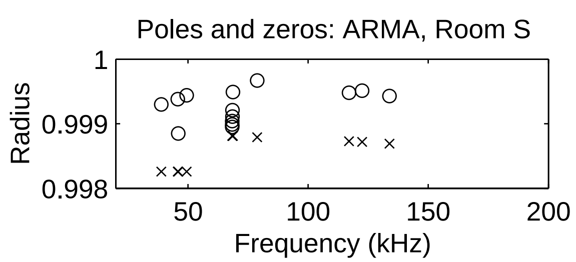

Fig. 8. Poles (×) and zeros (o) for the modal equalizer of ARMA (room S).

4.3. Selection of audio material for the test

Five test signals were chosen so that various signal types are covered (Table 8). Test signals 1-3 are 5 to 7-second clips of tracks 72, 30 and 5, respectively, taken from [15]. Signal 4 is a 0.5-second burst of white noise followed by silence, bandpass filtered at 20-200 Hz with a fourth-order Butterworth filter. Signal 5 is an impulse lowpass filtered twice at 200 Hz (first-order Butterworth) and once at 500 Hz (10th-order Butterworth), and in order to limit infrasound energy, first-order Butterworth highpass filter is applied at 27 Hz.

Plots of power spectra demonstrate the low-frequency energy content in the test signals (Fig. 12).

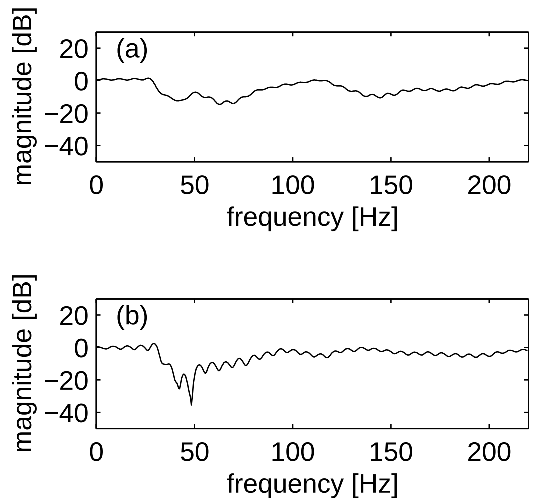

Fig. 9. Magnitude equalizer (Ampleq) responses, (a) magnitude responses for Room G and (b) for Room S.

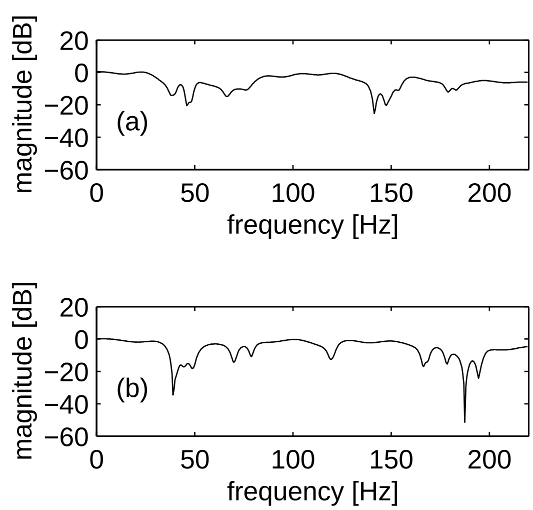

Fig. 10. Combined modal equalizer and magnitude equalizer responses (AMK), (a) magnitude responses for Room G and (b) for Room S.

4.4. Production of audio test items

The room impulse response measurements from the dummy head were used to produce listening test stimuli. Each dummy head impulse response measurement contains a binaural impulse response, i.e.

two responses, one for each ear in the dummy head. Both impulse responses were filtered with the equalizers to provide the equalized impulse responses.

The audio material for the subjective test was convolved with the four impulse response pairs (the original binaural room response, and the three equalized binaural room responses) for rooms G and S.

Stereophonic audio data was resampled to 44.1 kHz sampling frequency for convolution. Convolution processing of the audio material with binaural impulse responses was performed in floating point and the resulting data was rounded with dithering to 16 bit fixed point presentation.

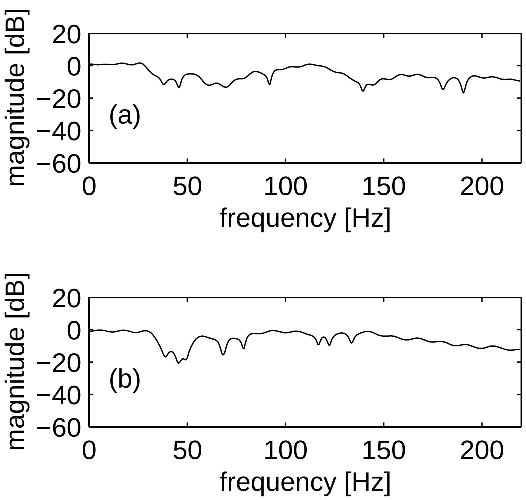

Fig. 11. Combined modal equalizer and magnitude equalizer responses (ARMA), (a) magnitude responses for Room G and (b) for Room S.

Table 8. Audio material for listening tests.

Sound	Description	Duration
1	Male voice	7 s
2	Drum hit 1	5 s
3	Drum hit 2	5 s
4	Bandpass filtered noise burst	5 s
5	Lowpass filtered impulse	5 s

4.5. Subjects

Temporal definition experiment had four subjects. The spectral balance experiment had three subjects. All subjects were male with normal hearing aquity, with ages ranging from 31 to 56 years. The listeners had earlier experience in listening to sound properties targeted by the study, and all had association to the research project. One of the subjects was the designer of the listening test.

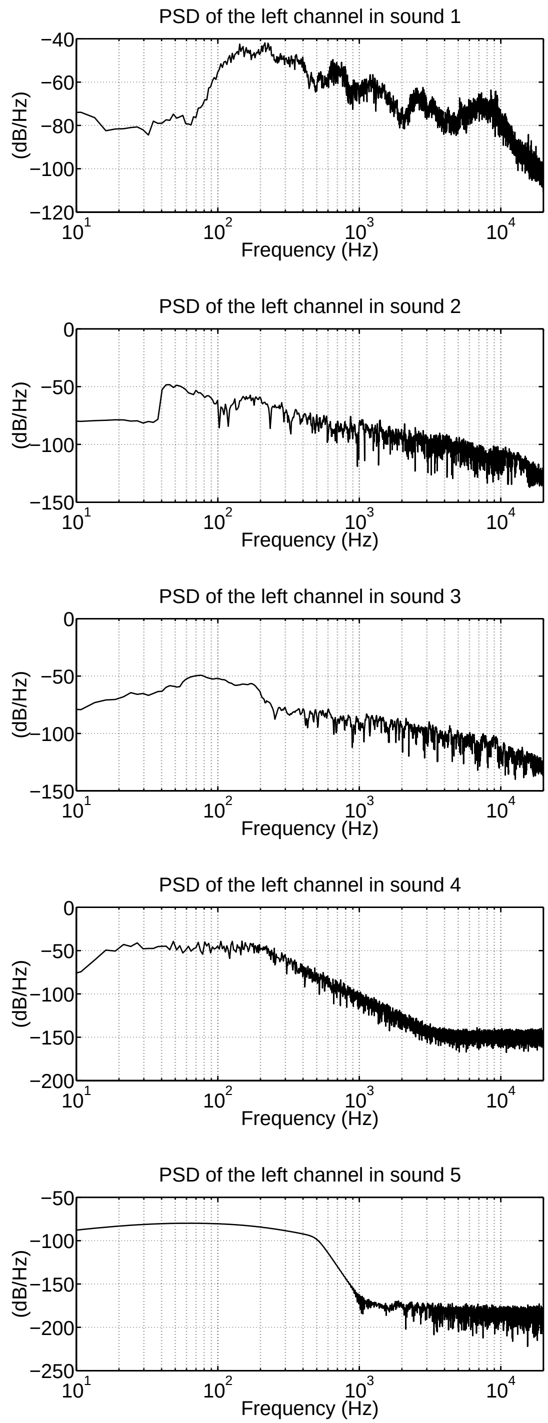

Fig. 12. Power spectral density plots of the test signals (left channel). Right channels spectra are not shown as they are very similar to the left channel spectra.

4.6. Method of assessment

Test signals were played binaurally through headphones using a subjective listening test automation software[2] [13,14]. Subjects were recommended a listening sound level, but allowed to initially adjust the listening level according to their preference with the recommendation to keep the sound level fixed then throughout the tests.

The task of the subjects was to listen to two audio signals A and B as many times as they wanted. The order, sequence and method of comparing the two signals were controlled entirely by the listener using a window-based user interface (Fig. 13). Repetition of items and change between items was allowed at any time and as many times as the subjects wanted. Subjects were to give a rating on a continuous grading scale between 0 and 10 for each signal using graphical sliders (Fig. 13). No verbal descriptors were given for values on the grading scale. The subjects were instructed to use the entire grading scale if possible. After rating of both items A and B clicking the "Done" button proceeded the test to the next pair of signals.

The data stored for each signal pair evaluation were the assessments with the precision of one decimal (three digit precision), the time taken to complete the evaluation (signified by clicking the “Done” button), the test items, the number of switches between the items, and the sequence number for the evaluation.

Subjects were asked to evaluate two qualities about the low-frequency reproduction of a pair of test items at a time. These were termed “temporal definition” and “spectral balance”. Prior to the test the subjects were given instructions, and all subjects discussed in a group with the instructor about the execution of the test task and the meaning of the qualities to be tested. A consensus opinion was reached of the properties assigned to each of the test objective qualities “temporal definition” and “spectral balance”. Subjects knew prior to the test that the target frequency range for testing was frequencies below 200 Hz. The subjects were asked to ignore possible high frequency differences in audio.

“Temporal definition” was agreed to signify the lack of detectable lengthening of the decay time at any discrete frequency within the test frequency range. It was recognized that such lengthening may also cause changes to the timbre of the low-frequency audio, but any timbre or coloration changes were to be ignored for this test item.

“Spectral balance” was agreed to signify the lack of coloration at any discrete frequency within the target frequency range. It was recognized that a coloration change may be associated with a lengthening of the decay at some frequency, but it was agreed that for this item any changes in the temporal characteristics were to be ignored.

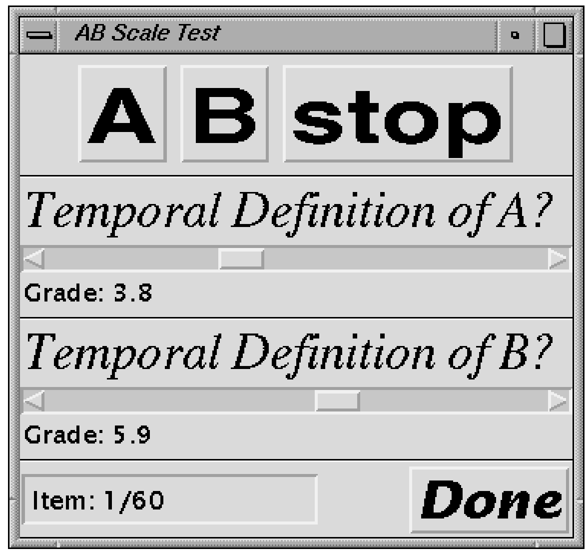

Fig. 13. Software user interface used in grading the subjective listening test items. Buttons “A” and “B” play the audio items to be graded. Grading is done by setting the two sliders.

4.7. Results

There were two room responses to be evaluated, termed Room S and Room G. Five audio test items were used. The original room response and the three processed alternatives were compared in a full permutation set. The processing alternatives for each room were magnitude equalization alone, modal equalizer design AMK with magnitude equalization, and modal equalizer design ARMA with amplitude equalization.

Each subject performed 60 pair-wise comparisons in the test. The median time of grading a signal pair was approximately 50 seconds.

The signal pairs (Table 9) were not randomized for each subject, for the audio test item or for the comparison order of processing alternatives, but the subjects did not have a chance to discuss the individual

[2] The listening test automation software used in the tests is called GuineaPig 2.0 and has been developed at the Helsinki University of Technology, Laboratory of Acoustics and Audio Signal Processing [13,14].

test items until all subjects had finished the test. The temporal property was assessed before the spectral property. Some learning during the test about the order in with the test item pairs were presented was reported.

Since no verbal descriptors for the grading scale values were given, the subjects used the grading scale in rather different ways. In order to enable pooling of data for different subjects, the grades given by each subject were scaled for the same mean and variance.

The Test results are presented in Figures 14-23. For room G, “G0” refers to the sound sample convolved with the room G response, “G1” magnitude equalized (Ampleq), “G2” to magnitude and modal equalized with a modal equalizer design using the AMK method and “G3” to magnitude and modal equalized, modal equalizer design using the ARMA method. Room S is marked similarly. The male speech and noise signals show lowest variance in gradings of the temporal definition (Fig. 14-18). The noise signal shows also the lowest variance for the gradings of the spectral balance (Fig. 19-23).

Table 9. Signal pairs and their presentation order. See Table A for equalizer abbreviations.

no	A	B
1	room	AMK
2	room	ARMA
3	room	Ampleq
4	ARMA	AMK
5	Ampleq	AMK
6	Ampleq	ARMA

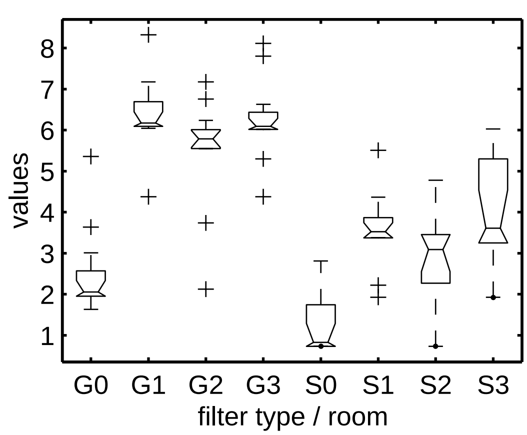

Fig 14. Gradings of “temporal definition” for sound sample 1.

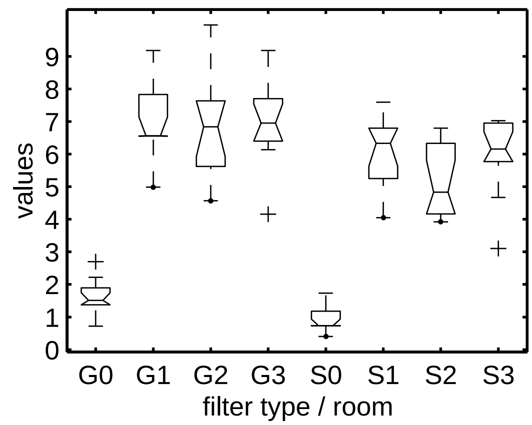

Fig 15. Gradings of “temporal definition” for sound sample 2.

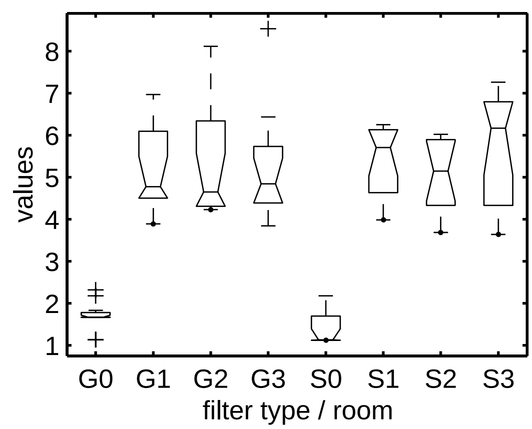

Fig 16. Gradings of “temporal definition” for sound sample 3.

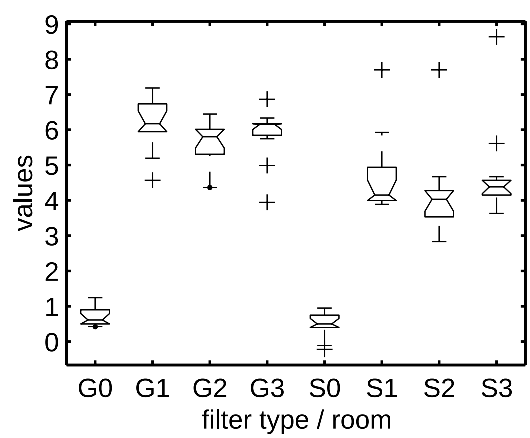

Fig 17. Gradings of “temporal definition” for sound sample 4.

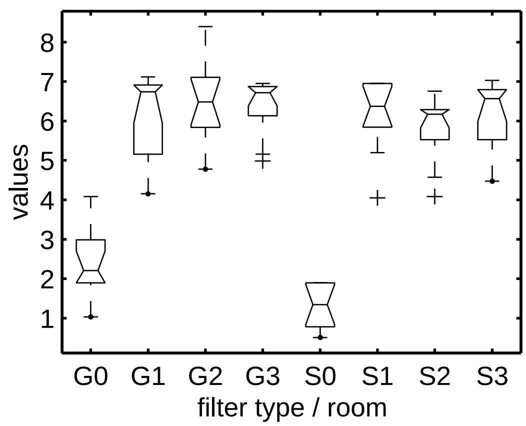

Fig 18. Gradings of "temporal definition" for sound sample 5.

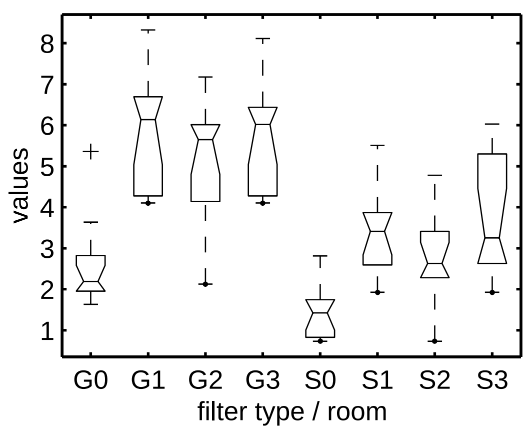

Fig 19. Gradings of "spectral balance" for sound sample 1.

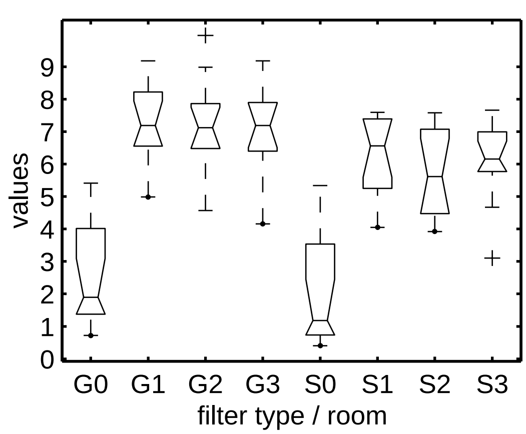

Fig 20. Gradings of "spectral balance" for sound sample 2.

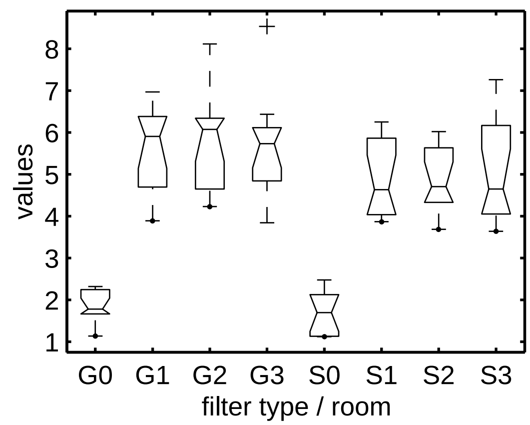

Fig 21. Gradings of "spectral balance" for sound sample 3.

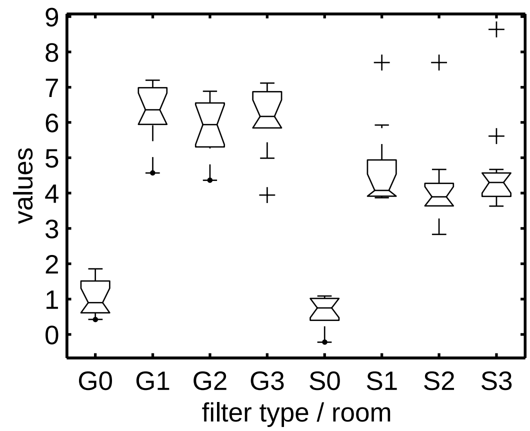

Fig 22. Gradings of "spectral balance" for sound sample 4.

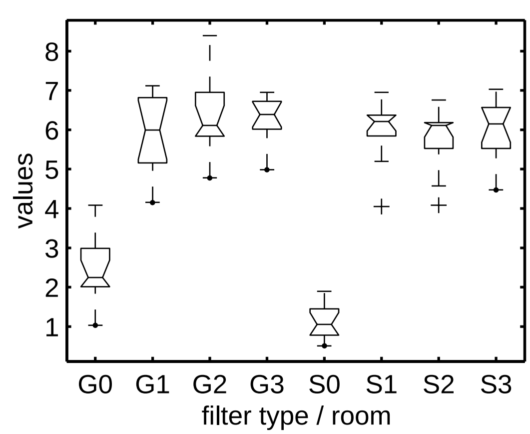

Fig 23. Gradings of "spectral balance" for sound sample 5.

4.8. ANOVA analysis

Matlab function `p = anova1(X)` [16] performs a balanced one-way analysis of variance for comparing the means of two or more columns of data in the *m*-by-*n* matrix *X*, where each column represents an independent sample containing *m* mutually independent observations. The function returns a probability level *p* for the null hypothesis that all samples in the data set *X* are drawn from the same population. A small value of *p* suggests that at least one sample mean in data is not from the same population as the other sample means.

ANOVA analysis was done to confirm statistical significance of differences in grade value means for each type of equalization. Tables 10 and 11 show the values of *p* when all four cases are included, including the unequalized sound. Tables 12 and 13 show the values of *p* when only equalized cases (Ampleq, AMK and ARMA) are considered.

Visual inspection of Figs. 14-23 suggests that unequalized test sounds received lower grades than equalized test sounds. Tables 10 and 12 confirm that the difference in mean is significant because *p* values are small. We can conclude that any one of the equalizations used produces a significant improvement. However, the incremental improvement of using also modal equalization along with magnitude equalization was not clearly demonstrated (Tables 11 and 13).

The gradings for the temporal and spectral properties are similar. This is in agreement with the fact that room modes typically cause an increase in the decay time as well as an increase in the room gain at that particular frequency. Since the testing was non-randomized between subjects the present data does not provide the possibility to assess the degree of orthogonality of the temporal and spectral properties.

When data for all audio samples is pooled together, the result does not change. Any form of equalization provides a significant improvement in both the temporal and spectral aspects of the low frequency reproduction. Since the third-octave smoothing based magnitude equalization is present in all equalization methods evaluated, although the actual magnitude equalizers differ for each equalization method, we can conclude that magnitude equalization provides a significant improvement in the quality of low frequency reproduction at frequencies below 200 Hz. The incremental improvement potentially provided by modal equalization remains unresolved.

Table 10. Value of *p* for temporal definition when grades of original and equalized signals are considered.

Sound	Room G	Room S
1	7.0e-11	4.0e-5
2	1.2e-13	3.1e-12
3	1.6e-11	8.1e-12
4	3.3e-16	5.9e-14
5	5.6e-11	0

Table 11. Value of *p* for temporal definition when grades of equalized signals (Ampleq, AMK and ARMA) are considered.

Sound	Room G	Room S
1	0.13	0.16
2	0.98	0.48
3	0.99	0.70
4	0.72	0.48
5	0.16	0.65

Table 12. Value of *p* for spectral balance when grades of original and equalized signals are considered.

Sound	Room G	Room S
1	0.29e-6	260e-6
2	450e-6	610e-6
3	21e-6	4500e-6
4	0.27e-6	4.1e-6
5	0.27e-6	7.7e-6

Table 13. Level of *p* for spectral balance when grades of equalized signals (Ampleq, AMK and ARMA) are considered.

Sound	Room G	Room S
1	0.55	0.35
2	0.89	0.91
3	0.70	0.90
4	0.33	0.67
5	0.23	0.99

5. DISCUSSION

5.1. Objective evaluation of AMK and ARMA methods

The objective evaluation demonstrated how both methods of designing modal equalizers work for sparsely placed modes, but that the AMK modal equalizer design method fails for closely spaced modes while the ARMA design method is able to deal with these cases better. Modal equalization is able to control the rate of the initial modal decay, but may fail to improve the decay rate at lower (later) decay levels.

5.2. Subjective listening test

The subjective listening test to assess the incremental improvement of wide-band magnitude equalization and modal equalization was unable to quantify the incremental improvement due to modal equalization. Informal comments from subjects suggest that some improvement was heard, but results suggest that third-octave magnitude equalization of low frequencies was able to produce similar level of improvement.

Although the subjective test lacked randomization and the number of subjects was small, it is clear that the incremental benefit of modal equalization along with magnitude equalization is significantly smaller than the incremental benefit of conventional magnitude response equalization to unequalized one. This statement cannot be interpreted to mean that modal equalization could not bring an audible improvement in situations where magnitude equalization of a listening space was to been done anyway. Such improvement may be beneficial for critical listening environments even when the incremental improvement is small.

It must be noted that the subjective listening test of this study is a preliminary one. Further investigations are needed, for example by using carefully controlled synthetic room modes, in order to find the limits to modal decay perception. On the other hand, longer-term comparison should be done and real music signals listened to instead of isolated sounds.

5.3. Conclusion

Modal equalization is a novel technique to specifically tailor decay properties of a listening space. In this paper we have demonstrated the efficiency of two different methods to design modal equalizers for low frequencies, and compared their incremental benefit to that of conventional magnitude equalization. Case studies demonstrated that the two methods differ in their ability to model closely spaced room modes. The equalizers designed on the basis of the real room measurements yielded two equalizer designs with measurably differing capacity to control modal behavior. Yet, in the subjective listening test, we could not demonstrate a significant incremental benefit of modal equalization over conventional magnitude equalization.

6. ACKNOWLEDGMENTS

This work is a part of the VÄRE technology program project TAKU (Control of Closed Space Acoustics) funded by the National Technology Agency of Finland. A part of the work was conducted in the SARA Project funded by the Academy of Finland (project number 201050). The authors are greateful to D.Sc.(Tech.) Hanna Järveläinen for helpful discussions on statistical analysis.

7. REFERENCES

[1] S. J. Elliott and P. A. Nelson, "Multiple-Point Equalization in a Room Using Adaptive Digital Filters," *J. Audio Eng. Soc.*, vol. 37, no. 11, pp. 899-907 (1989 Nov.).

[2] R. P. Genereux, "Adaptive Loudspeaker Systems: Correcting for the Acoustic Environment," in *Proc. AES 8th Int. Conf.*, (Washington D.C., 1990 May), pp. 245-256.

[3] J. Mourjopoulos and M. A. Paraskevas, "Pole and Zero Modelling of Room Transfer Functions," *J. Sound and Vibration*, vol. 146, no. 2, pp. 281-302 (1991).

[4] J. Mourjopoulos, "Digital Equalization of Room Acoustics," *preprint 3288, AES 92nd Conv.*, (Vienna, Austria), 1992 March.

[5] S. J. Elliott, L. P. Bhatia, F. S. Deghan, A. H. Fu, M. S. Stewart, and D. W. Wilson, "Practical Implementation of Low-Frequency Equalization Using Adaptive Digital Filters," *J. Audio Eng. Soc.*, vol. 42, no. 12, pp. 988-998 (1994 Dec.).

[6] A. G. Groh, "High-Fidelity Sound System Equalization by Analysis of Standing Waves," *J. Audio Eng. Soc.*, vol. 22, no. 10, pp. 795-799 (1974 Oct.).

[7] F. E. Toole and S. E. Olive, "The Modification of Timbre by Resonances: Perception and Measurement," *J. Audio Eng. Soc.*, vol. 36, no. 3, pp. 122-141 (1988 March).

[8] S. E. Olive, P. L. Schuck, J. G. Ryan, S. L. Sally, and M. E. Bonneville, "The Detection Thresholds of Resonances at Low Frequencies," *J. Audio Eng. Soc.*, vol. 45, no. 3, pp. 116-127 (1997 March).

[9] A. Mäkivirta, P. Antsalo, M. Karjalainen, and V. Välimäki, "Low-Frequency Modal Equalization of Loudspeaker-Room Responses," *Preprint 5480, AES 111th Conv.*, (New York, USA), 2001 Nov/Dec.

[10] M. Karjalainen, P. A. A. Esquef, P. Antsalo, A. Mäkivirta, and V. Välimäki, "AR/ARMA Analysis and Modeling of Modes in Resonant and Reverberant Systems," *Preprint 5590, AES 112th Conv.*, (Munich, Germany), 2002 May.

[11] M. Karjalainen, P. A. A. Esquef, P. Antsalo, A. Mäkivirta, and V. Välimäki, "Frequency-Zooming ARMA Modeling of Resonant and Reverberant Systems," *J. Audio Eng. Soc.*, vol. 50, no. 12, pp. 1012-1029 (2002 Dec.).

[12] M. Karjalainen, P. Antsalo, A. Mäkivirta, T. Peltonen, and V. Välimäki, "Estimation of Modal Decay Parameters from Noisy Response Measurements," *J. Audio Eng. Soc.*, vol. 50, no. 11, pp. 867-878 (2002 Nov.).

[13] J. Hynninen and N. Zacharov, "GuineaPig - a Generic Subjective Test System for Multichannel Audio," AES 106th Conv., Munich, (preprint 4871), 1999 May. See also http://www.acoustics.hut.fi/projects/GuineaPig2.

[14] J. Hynninen. "A software-based system for listening tests," *Master's Thesis*, Helsinki University of Technology, Espoo, Finland (2001). Available online at http://www.acoustics.hut.fi/publications/.

[15] "Dynamic Sound Power Test," Blue Flame CD 1, Gema 1989 Germany, LC 8593, no 40212.

[16] The Mathworks, Inc., "Matlab 6.1 Statistical Toolbox User Manual," Natick, Massachusetts, 2001.

APPENDIX 1

Table 14. Poles and zeros for the modal equalizer of AMK (room G).

Frequency (Hz)	Zero radius	Pole radius
47.866	0.99956	0.99843
47.603	0.99964	0.99843
148.109	0.99902	0.99843
40.120	0.99933	0.99843
154.236	0.99924	0.99843
142.184	0.99921	0.99843
66.823	0.99873	0.99843
66.693	0.99880	0.99843
46.534	0.99963	0.99843
147.784	0.99918	0.99843
141.815	0.99934	0.99843
38.061	0.99923	0.99843
66.441	0.99881	0.99843
149.862	0.99921	0.99843
76.873	0.99860	0.99843
183.526	0.99902	0.99843
66.292	0.99882	0.99843
76.807	0.99862	0.99843
76.614	0.99865	0.99843
179.168	0.99912	0.99843

Table 15. Poles and zeros for the modal equalizer of AMK (room S).

Frequency (Hz)	Zero radius	Pole radius
48.980	0.99939	0.99843
40.413	0.99950	0.99843
44.650	0.99921	0.99843
40.386	0.99954	0.99843
78.930	0.99955	0.99843
48.922	0.99886	0.99843
70.260	0.99921	0.99843
48.857	0.99912	0.99843
40.135	0.99959	0.99843
119.194	0.99922	0.99843
70.154	0.99938	0.99843
195.960	0.99936	0.99843
166.308	0.99935	0.99843
168.947	0.99930	0.99843
178.539	0.99935	0.99843
187.518	0.99938	0.99843
44.589	0.99916	0.99843
188.814	0.99937	0.99843
195.196	0.99932	0.99843
120.688	0.99926	0.99843

Table 16. Poles and zeros for the modal equalizer of ARMA (room G).

Frequency/Hz	Zero radius	Pole radius
38.040	0.99931	0.99826
45.901	0.99962	0.99826
92.127	0.99965	0.99877
139.680	0.99941	0.99868
145.465	0.99887	0.99867
180.368	0.99922	0.99860
190.894	0.99928	0.99857

Table 17. Poles and zeros for the modal equalizer of ARMA (room S).

Frequency/Hz	Zero radius	Pole radius
38.918	0.99930	0.99826
45.748	0.99938	0.99826
45.992	0.99885	0.99826
49.473	0.99944	0.99826
68.445	0.99895	0.99881
68.462	0.99899	0.99881
68.484	0.99904	0.99881
68.513	0.99911	0.99881
68.555	0.99921	0.99881
68.768	0.99949	0.99881
78.830	0.99967	0.99879
117.056	0.99948	0.99873
122.510	0.99951	0.99872
133.870	0.99943	0.99869