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Virtual acoustics in practice rooms

Master's Thesis submitted in partial fulfillment of the requirements for the degree of Master of Science in Technology.

Espoo, Oct 15, 2007

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HELSINKI UNIVERSITY OF TECHNOLOGY	ABSTRACT OF THE MASTER'S THESIS

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Due to their small size, the acoustical design of music practice rooms at music institutes presents designers with challenges and requires compromises between absorption and reverberation. In practice, some reverberation is needed to provide support and more lively sound for the musician. Playing in a room with plenty of absorption and very little reverberation is often uninspiring and easily causes fatigue. On the other hand, increasing reverberation by reducing the absorption results in higher sound pressure levels. As music teachers are exposed to these conditions for long periods of time, in the long term this can lead to hearing problems. Similar problems are often also faced by orchestras when rehearsals are held in facilities with poor acoustical conditions.

This work investigates the use of an electro-acoustic system to improve the acoustics in practice spaces. The electro-acoustic system uses an active method to create additional reverberation without increasing the sound pressure levels. A second objective is to evaluate the acceptance of virtual acoustics by users in music practice spaces.

For this purpose, a prototype of an electro-acoustic system was installed in three different rooms, and the reverberation time and sound pressure level of the rooms were measured. These case studies consist of two small classrooms and a theater hall converted to a rehearsal hall for a symphony orchestra. For each case study, subjective opinions were collected. The results for each room were then analyzed in terms of the feedback from the users. The case studies show that with a careful tuning, and especially in larger spaces, a virtual acoustic system is a potential solution for improving the room acoustics of a practice room.

Keywords: electro-acoustics, practice room, rehearsal hall, reverberation enhancement, room acoustics, time variance, truncated hall

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Musiikin harjoitusluokkien pienestä koosta johtuen akustiikkasuunnittelussa joudutaan usein tekemään kompromisseja jälkikaiunnan ja tarvittavan vaimennusmateriaalin määrän suhteen. Harjoittelua ja instrumentin ääntä tukevaa sointia silmälläpitäen olisi suotavaa, ettei huone olisi kaiunnaltaan täysin kuiva. Kaiuttomassa tilassa soittaminen käy nopeasti raskaaksi ja saattaa vähentää aloittelevien soittajien harjoitteluintoa. Jälkikaiuntaa pieneen tilaan tavoiteltaessa joudutaan perinteisin akustisin menetelmin vähentämään vaimennuksen määrää, mikä puolestaan nostaa tilan äänitasoa. Varsinkin musiikin opettajien pidemmät altistusajat hyvin kaiuntaisissa opetusluokissa saattavat johtaa kuuloon liittyviin ongelmiin. Orkesterien tilanne on usein samankaltainen, kun harjoituksia joudutaan varsinaisen konserttisalin sijasta järjestämään akustisesti sopimattomissa tiloissa.

Tässä työssä tutkitaan sähköakustisen järjestelmän käyttöä harjoitustilojen akustiikan parantamiseksi. Tarkoituksena on näin ollen lisätä jälkikaiuntaa nostamatta tilan äänitasoa. Tavoitteena on myös tutkia käyttäjien suhtautumista sähköakustiikan käyttöön.

Työssä esitettyjä tutkimuksia varten sähköakustista järjestelmää kokeiltiin yhteensä kolmessa tilassa. Tutkituista tiloista kaksi ensimmäistä olivat tyypillisiä musiikkioppilaitoksen pieniä harjoitus- ja opetusluokkia. Kolmas tila oli monikäyttösali, johon sähköakustiikan avulla pyrittiin luomaan konserttisalin lavaa muistuttava akustiikka sinfoniaorkesterin harjoitusta varten. Tiloissa tehtiin huoneakustisia mittauksia sekä haastateltiin käyttäjiä kyselytutkimuksella. Tutkimusten tulosten perusteella on todettavissa, että harjoitustilojen akustiikan parantaminen sähköakustisella järjestelmällä on mahdollista.

Avainsanat: sähköakustiikka, harjoitushuone, saliakustiikka, jälkikaiunta, huoneakustiikan parantaminen, aikavarianssi, katkaistu sali

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Jukka Pätynen

Contents

Symbols

α	Absorption coefficient
A	Absorption area
c	Speed of sound in air
β	Maximum phase deviation
B	Bandwidth
f_g	Schroeder frequency
G_{ML}	Gain matrix
$h(t)$	Acoustical impulse response
H	Transfer function
$\mathbf{I}$	Unity matrix
k	Wave number
λ	Wave length
m	Air absorption coefficient
$m(t)$	Modulation function
n_l	Number of loudspeakers
n_m	Number of microphones
φ	Incident sound angle
p	Pressure
$R(\omega)$	Received signal in frequency domain
$\theta(t)$	Phase angle
V	Room volume
S	Surface area
$S(\omega)$	Source signal in frequency domain
$x(t)$	Input signal

Abbreviations

ACS	Acoustic Control System
AD/DA	Analog-to-digital / digital-to-analog conversion
AR	Assisted Resonance
DM	Delay modulation
DSP	Digital signal processing
EDT	Early decay time
EDTF	Early decay time frequency ratio
EDTP	Early decay time on platform
FIR	Finite impulse response
FM	Frequency modulation
FS	Frequency shift
GBI	Gain before instability
IIR	Infinite impulse response
LTI	Linear, time-invariant
LTV	Linear, time-variant
MIDI	Musical instrument digital interface
MLG	Mean loop gain
MLS	Maximum length sequence
PM	Phase modulation
RT_{60}	Reverberation time
SNR	Signal-to-noise ratio

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Chapter 1

Introduction

In all musical environments, finding a suitable acoustic space is an important aspect for consideration. The variety of different instruments and ensembles presents different requirements for optimal acoustics, whether a rehearsal or performance is in question. Especially in buildings constructed without modern standards and specifications, the practice conditions can be far from optimal, since the properties of the rooms are usually inadequate to be natural sounding and acoustically prepared for efficient rehearsing. Still, the musical education is becoming more popular, thus increasing the number of students in music schools, hence demanding a greater number of good practice rooms where the playing is pleasant and inspiring. Small rooms for music teaching can for instance be too small or feature too loud reverberation to offer a relaxing environment for a teacher and a pupil.

Orchestras are often facing a similar situation. For them, the optimal rehearsal space is the hall in which they are giving performances. Since the concert halls are increasingly designed to be multi-purpose halls suitable for instance drama, dance, and film presentations in addition to classical music, the rate of use is getting higher. This leaves orchestras less time to rehearse in their characteristic performance environment and making it harder to adopt a suitable musical interpretation.

These problems point out a growing need for good practice spaces. As renovating existing rooms can be expensive, other methods of improving the acoustics are often preferred. Previous studies have shown that with modern signal processing techniques, it is possible to extend the acoustical volume of a room with a configuration consisting of loudspeakers, a small number of microphones, standard computer equipment and in most cases slight room acoustical modifications. Compared to rebuilding, such *virtual acoustic* method is an attractive alternative, since it can be much less expensive, the installation requires less work and in addition, it is capable of creating variable acoustic conditions instead of a single fixed one. Even a room originally unintended for music practice can be converted into a proper practice space with virtual acoustic solution, provided some further modifications to the room are feasible.

The first published research on *electro-acoustic* systems for reverberation enhancement dates from over 30 years ago. During the last ten years, virtual acoustics has gathered more attention as the required technology has become more affordable. A small number of commercial concepts have been developed since by specialized manufacturers. While most of the products are intended for either enhancing reverberation in existing halls, only few are designed for practice room use. A master's thesis by Nummela [48] presented an implementation of a prototype system that could provide a virtual acoustic environment for full-scale symphony orchestras. Such a system with minor adaptations is used in this work to investigate the usage of virtual acoustic system in practice. Theoretically, an electro-acoustic system could be refined to such a level that modeling of existing spaces is possible. However, an accurate modeling of a specific hall or room demands a lot of work, because this system depends on traditional room acoustics in addition to the electro-acoustic system.

1.1 Research problem statement

The purpose of this work is to investigate whether virtual acoustics can be used in practice to enhance or improve room acoustics in existing music practice spaces of different sizes. For this purpose, three real world installations are studied. The methods consist of analyzing objective acoustical measurements and subjective opinions from the real world installations.

Chapter 2

Room acoustics

Sound in every space is a mixed combination of direct sound waves emanating from the source and reflected waves from the surroundings of an environment. Depending on the nature of boundaries in conventional acoustical space, the distribution of audible sound between direct sound and its reflections varies greatly. This variation creates the fundamental acoustical features in rooms.

Traditionally, the acoustic qualities of a room have been a result of a careful combination of forms, materials and structures. These architectural matters affect the direction and level of the reflections which in proper mixture forms a favorable acoustical environment for performances and practice in halls and rooms. Because the modern understanding of room acoustics is based on the importance of the relation between the direct sound, the first reflections and the reverberant sound, the design and analysis of the boundaries are taken under careful investigation when designing or improving acoustical spaces [42, 31]. Various different approaches to the acoustical analysis exist, of which some of the principal theories are presented here.

2.1 Sound propagation in spaces

Sound in all spaces can be seen as pressure waves propagating longitudinally with a certain frequency and amplitude in a medium from a single point or a set of sources. With the basic assumption of a homogeneous medium, the propagation is straightforward. Changes in pressure are caused by the movement of particles in a compressible fluid. For a simple sinusoidal wave, the displacement of particles [72] in two-dimensional cartesian coordinates follows the equation

$$y(x,t) = A\sin(\omega t - kx)\ . \tag{2.1}$$

Here the sound wave has a direction $+x$ and at a certain moment of time t, it has the displacement y. $k = \omega/c$ is called *wave number* and parameter $\omega = 2\pi f$ is a function of the frequency f. c denotes the speed of sound. By substituting the particle velocity using the laws of conservation of momentum and compression, the equation can be expressed as differential equation

$$\frac{\partial^2 p}{\partial x^2} = \frac{1}{c^2}\frac{\partial^2 p}{\partial t^2} \tag{2.2}$$

with pressure p as function of time and place [31]. An expression of this differential equation is possible so that the wave propagation can be inspected in a three-dimensional space:

$$\nabla^2 p = \frac{1}{c^2}\frac{\partial^2 p}{\partial t^2}. \tag{2.3}$$

Equation (2.3) is called the *Helmholz equation* [72]. It is noticeable that equation (2.1) is an arbitrary solution for the differential equation above [31], [72].

Having the acoustic space limited by rigid walls, boundary conditions must be set for wave equations. For simplicity it is assumed - dissimilarly to natural phenomena - that incident waves are plane waves. Calculating reflections for natural sound waves propagating spherically is more complicated. The reflection factor of a boundary is defined by the change in the amplitude and phase. Complex reflection factor $R = |R|e^{i\varphi}$ is used to express *absorption coefficient* of a wall [31]:

$$\alpha = 1 - |R|^2. \tag{2.4}$$

The sound absorption in more comprehensively reviewed later in this chapter. In addition to the absorption coefficient, *acoustic impedance* must be defined for describing the acoustic properties of a wall as following [31]:

$$Z = \frac{p}{v_n}, \tag{2.5}$$

where v_n denotes the component of the particle velocity normal to the wall surface. Since an ideal, non-absorbing surface has zero particle velocity component normal to the wall, the impedance is evaluated as infinite, thus leading to boundary conditions

$$\frac{dp_x}{dx} = 0, \quad x = 0 \vee\ x = L_x \tag{2.6}$$

$$\frac{dp_y}{dy} = 0, \quad y = 0 \vee\ y = L_y \tag{2.7}$$

$$\frac{dp_z}{dz} = 0, \quad z = 0 \vee\ x = L_z\ . \tag{2.8}$$

These conditions are for ideal rectangular room with dimensions L_x, L_y and L_z. In this case, the wave equation is of form

$$\frac{\partial^2 p}{\partial x^2} + \frac{\partial^2 p}{\partial y^2} + \frac{\partial^2 p}{\partial z^2} + k^2 p = 0 \quad , \tag{2.9}$$

where k denotes a variable of characteristic value. With initial and boundary values in effect, performing a separation of variables is possible, thus being able to have the form

$$p(x,y,z) = p_x(x)p_y(y)p_z(z) \Rightarrow \begin{cases} \frac{d^2 p_x(x)}{dx^2} + k_x^2 p_x = 0 \\ \frac{d^2 p_y(y)}{dy^2} + k_y^2 p_y = 0 \\ \frac{d^2 p_z(z)}{dz^2} + k_z^2 p_z = 0. \end{cases} \tag{2.10}$$

and having also

$$k_x^2 + k_y^2 + k_z^2 = k. \tag{2.11}$$

The solution for equation (2.10) is of form

$$p_x(x) = A_x \cos(k_x x) + B_x \sin(k_x x) \tag{2.12}$$

correspondingly for all axis. Constants A_x and B_x are determined against boundary conditions (2.6), and therefore

$$\frac{dp_x}{dx} = 0, \quad \text{when } x = 0 \Rightarrow B_x = 0 \tag{2.13}$$

$$\frac{dp_x}{dx} = 0, \quad \text{when } x = L_x \Rightarrow k_x = n\pi/L_x, n \in 1, 2 \dots . \tag{2.14}$$

k_y and k_z are solved correspondingly. Placing constants from equation (2.13) for x, y and z into equation (2.11) eigenvalues for wave equation

$$k = \pi\sqrt{(\frac{n_x}{L_x})^2 + (\frac{n_y}{L_y})^2 + (\frac{n_z}{L_z})^2} \tag{2.15}$$

and room eigenfrequencies

$$p(x,y,z) = C\cos(\frac{n_x \pi x}{L_x})\cos(\frac{n_y \pi y}{L_y})\cos(\frac{n_z \pi z}{L_z}) \tag{2.16}$$

$$f_{n_x n_y n_z} = \frac{k_{n_x n_y n_z} c}{2\pi} \tag{2.17}$$

can be solved [31]. With different values for $k = 1, 2 \dots$ room modes of k th order are calculated, of which the lowest ones are important in small room acoustic design principles. In a space having only small differences between the dimensions L_x, L_y and L_z, it is noticeable that the room modes are also distributed proportionally to the corresponding dimensions or its multiplies [57]. To avoid disturbing effects in small rooms, it is recommended that the dimensions are carefully selected to obtain evenly distributed modes. Examples of certain favorable proportions can be found in [66] and [40].

The above solution for the wave equation is directly applicable only to perfectly rectangular rooms with ideal reflective walls which in real world does not exist. Solving the equations for surfaces with non-infinite impedance is possible, unlike for arbitrary shapes, which is no longer sensible. Numerical techniques, such as finite element method (*FEM*), finite volume method (*FVM*) and boundary element methods (*BEM*) are developed for solving wave equation in more complex domains. The use of FEM method has been studied by i.e. Pietrzyk and Kleiner [52] while more traditional modeling relying on geometrical methods have been reviewed by Kuttruff [31] as well as Svensson and Kristiansen [63].

2.1.1 Absorption

By the term of improving acoustics in small rooms, the unprofessional understanding is that the process is about clapping hands at different positions and installing proper convenient panels to the walls. In more scientific sense, it is not so far-fetched than it sounds like. Studying existing sound fields and treating the surfaces with appropriate materials describes perhaps better the room acoustical methods. In this section, principles of common absorption materials and their effect to the acoustical conditions are discussed.

In rooms where the dimensions and wall proportions cannot be altered, themeans for controlling reflections are in most cases limited to the use of absorptive and diffusing materials. Unwanted reflections can be attenuated to some extent with such elements, depending on the original state and the intended result. In the case of a specular reflection from a hard surface, placing an absorbent results in a reduced amplitude of the reflected sound. Choosing a suitable absorbent depends on the frequency band that is wanted to attenuate.

There are mainly three types of absorbents used, of which each is suitable for absorbing sound energy at a certain frequency band. In porous materials, such as carpets and acoustical panels, the sound absorption is effective from low middle frequencies and reaches the highest effect at high frequencies. Physically the sound absorption occurs in the numerous pores in the material in which the energy is converted into heat due to the increased friction of particle movement [57]. Sudden changes in the direction of movement increases the effect [64]. As the absorption is most effective when the amplitude maximum of the wave is placed inside the material, porous materials of equal thickness function more effectively when they are suspended from the ceiling or walls. The resulting air gap improves the absorption compared to a situation where the material is directly attached to the surface. From this phenomenon it is possible to calculate the lower frequency limit of effective absorption, although the thickness of magnitude $\lambda/4$ is not required for noticeable absorption [64]. Wave length is denoted with λ.

A second type of absorption materials is panel absorbents, whose effect is based on membrane resonance typically present in wall structures. Common wall types featuring good absorption are light partition walls and doors - generally all light, larger surfaces that are comparably loosely fastened. A fairly good example is a basic gypsum board wall on a wooden frame. The effect of resonant absorption can be further increased by combining it with a porous absorbent placed beneath the surface. Unlike porous materials, panel absorbents are most effective at low frequencies, depending on the mass of the material and the method of attachment [31].

A third kind of absorbing element is Helmholz resonator, which differs from the other two absorbers by being specially tuned for a limited frequency band. Usually this type of absorption is used in small rooms for reducing energy at low frequencies, which are often causing problems due to the low frequency modes [57]. The portability of such element is an advantage when a single room requires different absorption characteristics.

In Table 2.1, the absorptive properties of selected materials are presented by octave bands [57]. These materials are mainly utilized in the case studies further in this work. In the table, it is noticeable between the first two rows, how a treatment alters the absorption propeties in certain materials. According to Rossing [57], a plain concrete works fairly well as a wide-band absorbent, but with paint, the porous surface features are covered, thus reducing the effect especially above 250 Hz.

Table 2.1: Absorption coefficients of selected materials in octave bands.

	125	250	500	1000	2000	4000 Hz
Concrete block, unpainted	0.36	0.44	0.31	0.29	0.39	0.25
Concrete block, painted	0.10	0.05	0.06	0.07	0.09	0.08
Glass, window	0.35	0.25	0.18	0.12	0.07	0.04
Drapery, heavyweight	0.14	0.35	0.55	0.72	0.70	0.65
Wood floor	0.15	0.11	0.10	0.07	0.06	0.07
Acoustical tile, on concrete	0.14	0.20	0.76	0.79	0.58	0.37
Gypsum board, 12 mm	0.29	0.10	0.05	0.04	0.07	0.09

Instead of purely absorbing the sound energy in reflections, the alternative of using diffusing elements is in some cases justified. In rooms where the smooth, parallel walls have low absorption coefficients and otherwise a certain level of liveliness is required, installing diffusors to divide reflected sound energy more evenly into all directions is beneficial. Everyday elements featuring diffusing properties are for instance bookshelves and armchairs, although specially designed diffusor panels are usually preferred in more professional environments. Correctly placed diffusors result in a pleasant, less colored sound field [57]. Diffuse sound fields are more closely discussed in Section 2.1.2.

The methods discussed here for altering absorption properties of a space give useful means to control acoustical properties, whether by simply adding beneficial materials or modifying the wall structures. In existing rooms, the use of acoustical panels and other easily installable absorbents are usually favored, since altering wall structures is often considered too extensive operation and is regarded the last resort in difficult cases.

2.1.2 Diffuse sound field

Theory of diffuse sound field is a common assumption used with the investigations of the behavior of reverberant sound in closed spaces. The main idea behind the concept is that the flow of sound energy is statistically zero, thus the sound intensity is also zero. In practice this means that in a diffuse field, reflections from surrounding surfaces are equal from all random directions. An important property can be derived from the diffusion assumption. When the attenuation of direct sound from a source naturally follows the $1/r$-rule, thus halving the sound pressure when the distance is doubled, in diffuse field the attenuation is zero.

To define more accurately where this rule is valid, the boundary for diffuse sound field must be established. A property of diffuse field in closed space called *reverberation radius* is derived from the equation of pressure levels of direct and reverberant sound:

$$L_p = L_w + 10\lg\left(\frac{Q}{4\pi r^2}\right), L_{p,reverberant} = L_w + 10\lg\left(\frac{4}{A}\right) \Rightarrow \tag{2.18}$$

$$L_p = L_w + 10\lg\left(\frac{Q}{4\pi r^2} + \frac{4}{A}\right) \Rightarrow \tag{2.19}$$

$$\frac{Q}{4\pi r^2} = \frac{4}{A} \Leftrightarrow \tag{2.20}$$

$$r = \frac{1}{4}\sqrt{\frac{A}{\pi}}\,, \tag{2.21}$$

where A is the absorption area, similarly as in the equations for calculating reverberation time. The result r obtained from the equation is the minimum distance from a single source where the sound field is supposed to be diffuse. In this case omnidirectional source ($Q = 1$) is supposed. With other directivity values the equation changes respectively.

In special cases where absorption is not evenly distributed on the surfaces, the equation (2.21) gives only a coarse approximation, which results from the non-diffuse property of the sound field. In such situation the sound intensity is not zero in the vicinity of the more absorbing surfaces, thus the original assumption of diffuse field is not valid.

In addition a low-frequency boundary, over which frequencies the room can be supposed to be diffusing enough, can be evaluated by the Schroeder frequency:

$$f_g = 2000\sqrt{\frac{RT_{60}}{V}}, \tag{2.22}$$

where V represents the total volume of the room. RT_{60} denotes the reverberation time (see definition in Section 2.2.1). The principle behind this theory is to approximate in which frequencies the room modes (equation 2.17) begin to lose their resonant characteristics. Below this threshold the frequency response measurements can give erratic results if the receiving positions are carelessly selected. However, in large halls this is a negligible issue, since these frequencies are below normally measured range.

In generic small rooms, sound diffusion can be improved by adding arbitrary objects with sufficient dimensions. An effective diffusion is possible with additional obstacles that have minimum measures comparable to the wavelength. The importance of diffusing elements is emphasized in small listening spaces such as sound studios.

The importance of diffuse sound field is the most apparent in the definition of reverberation time, since the definition of RT_{60} assumes that the reverberant sound field is fully diffuse [57]. In a situation where a considerable amount of absorptive materials is concentrated on a single area, sound intensity is no longer zero, thus the flow in sound energy has a direction toward the absorbing surface. This breaks the condition of diffuse sound field, but according to Nummela [48], the result from reverberation radius calculation by using equation (2.21) can be regarded as a reference, and the real reverberation radius can be assumed higher than the resulting number.

Since the absorption area depends on absorptive properties of all surfaces, it is noticeable that in cases where active acoustics is used together with additional absorption, the reverberation radius does not remain constant, thus having an influence on microphone and loudspeaker positioning. On the other hand, the reverberation radius of instruments and loudspeakers must be considered in active acoustic system design in order to gain balanced sounding results and avoiding unwanted localization effects.

2.2 Room response

Natural acoustical systems in closed spaces are usually assumed to have properties of a linear, time-invariant (LTI) system. Conditions that are to be fulfilled are linearity, stability, time-invariant parameters and causality, of which linearity sets additivity ($f(x + y) = f(x) + f(y)$) and homogeneity ($f(ax) = af(x)$) properties. Stability requires that the response from system to an input signal is limited, whereas time-invariance demands that parameters of the system response are constant through time. Causality condition supposes that the system produces no response before an input signal is supplied [16].

Some of the conditions may be broken in unrestricted spaces or by an active acoustics. Common examples of such cases is the varying wind in open areas, which changes the overall system response depending on time. With an electro-acoustic system, feedback resulting from poorly designed algorithms or adversely positioned microphones or loudspeakers can break the stability property by generating a response saturated to the maximum amplification limit. However, some system implementations have time-variant features specifically in order to prevent instability as presented later in this work.

The impulse response $h(t)$ results in room acoustics from the direct sound and the numerous reflections. The complete behavior of sound in that space can be analyzed from the impulse response, although monaural response does not completely describe how a sound in certain space is heard. In Figure 2.1, a schematic impulse response from a small hall is illustrated. After the direct sound there is a group of early reflections, and after that the reflections occur so rapidly at the reverberation (truncated in the figure), in which case studying single reflections separately is not sensible. In smaller spaces late reverberation begins earlier than the 100 ms marked in the figure. The division between early reflections and reverberation depends also on psychoacoustical phenomena, which are discussed in Section 3.6.

Depending on the reflective properties of the room, in most rooms and halls the direct sound combined with early reflections form the greatest part of total received sound energy. Temporal sections shown in the figure are generalized approximates, since the impulse response depends on the distance from a source to the receiver as well as on the dimensions of the room [4]. The impulse response - or a time-energy curve in this case - should also be preferably smooth by its envelope. Deviations like echoes in otherwise gradually attenuating reflections are easily perceived artifacts.

Mathematically, the effect of the room response $h(t)$ to the sound from the source $x(t)$ is presented as a convolution

$$y(t) = \int_0^\infty h(\tau)x(t-\tau)d\tau, \tag{2.23}$$

where τ is used as convolution variable [16]. Respectively in frequency domain, the frequency response is obtained with a multiplication $Y(\omega) = H(\omega)X(\omega)$, where room frequency response $H(\omega)$ is derived with Fourier transform

$$H(\omega) = \int_0^\infty h(\tau)e^{-j2\pi ft} \tag{2.24}$$

for causal systems ($h(t) = 0, t < 0$) [16].

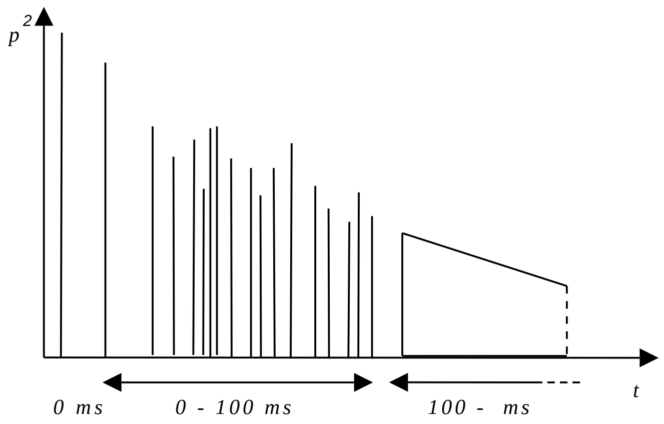


Figure 2.1: A basic model for impulse response in closed spaces.

2.2.1 Reverberation time

The measure of reverberation time is the most commonly used room acoustical parameter. The definition of reverberation time is the time in which the sound power level drops 60 dB from the original power level after the sound source is turned off. RT_{60} is the notation used for reverberation time in this work. Desirable values for RT_{60} are approximately 0.5-1.0 s for speech and 1.5-2.5 s for orchestral music [57]. An appropriate reverberation time for each purpose depends on the nature of the performance as well on the size of the hall. Larger spaces tend to naturally feature longer reverberation since the surface materials are fairly generally used in auditoriums independently of the dimensions.

The simplified classic equation for calculating reverberation time was formulated by W.C. Sabine [57]:

$$RT_{60} = 0.163\frac{V}{A}\text{ , where } A = \sum_{i=1}^{n} S_i\alpha_i. \tag{2.25}$$

$$RT_{60,air} = 0.163\frac{V}{A+4mV}. \tag{2.26}$$

In the equation above V represents the total volume of the space and S_i the area of each surface having absorption coefficient of α_i. The constant in the equation is for metric values, and it is empirically defined. Due to the frequency-dependent absorption coefficient, the reverberation time obtained from the Sabine equation must be evaluated separately for each frequency band to have more comprehensive results.

The equation (2.25) does not take the absorption of air into account. In large spaces, substitution $A_{air} = A + 4mV$ in the equation (2.26) can be made to include the effect of air absorption [31], where the air absorption is noted with m. The effect is notable only at frequencies above 2000 Hz.

However, the equation (2.25) has some properties that are inconsistent with the practice. For anechoic rooms, this equation gives a result of existing reverberation. Therefore more accurate mathematical approaches have been developed to replace the subjectively devised Sabine equation. Kuttruff [31] has stated that the total energy of reflected sound at a certain time t in a room is

$$E(t) = E_0 \exp\{[-mc + \overline{n}\ln(1-\overline{\alpha})t]\}, \tag{2.27}$$

where c represents the sound velocity in air, E_0 the initial sound energy, m the absorption of air, $\overline{n}$ the average count of reflections occurring per second and $\overline{\alpha}$ the average absorption coefficient of all surfaces of the room, thus

$$\overline{\alpha} = \frac{1}{S}\sum_i S_i\alpha_i. \tag{2.28}$$

In the equation above S is the total area of all surfaces of the room. Variable $\overline{n}$ in equation (2.27) can be solved with equation

$$\overline{n} = \frac{c}{2}\left(\frac{1}{L_x} + \frac{1}{L_y} + \frac{1}{L_y}\right) = \frac{cS}{4V}, \tag{2.29}$$

where L_x, L_y and L_z are the dimensions of the room. By combining equations (2.29) and (2.27)

$$E(t) = E_0 \exp\left(-ct\left[\frac{4Vm - \ln(1-\overline{\alpha})t}{4V}\right]\right). \tag{2.30}$$

Reverberation time RT_{60} representing the sound energy of one millionth of the original, substitution $E(t) = E_0\ 10^{-6}$ can be made, thus giving

$$t = RT_{60} = \frac{24V\ln 10}{4mV - A\ln(1-\overline{\alpha})} \tag{2.31}$$

$$= -0.163\frac{V}{A\ln(1-\overline{\alpha}) - 4mV}. \tag{2.32}$$

The latter equation is known as the *Eyring reverberation formula*. By assuming that $\overline{\alpha}$ is small, substituting $-\ln(1-\overline{\alpha})$ with a series expansion and neglecting the terms higher than second order, equation (2.31) is equal to the Sabine reverberation formula with air absorption included (equation 2.26). Comparative evaluations between Eyring and Sabine formulas have been performed in measurements of large churches by Carvalho [12].

In addition to the two formulas for reverberation time calculation, probably the next familiar method is the Fitzroy equation [19], which is recommended to be used in cases where the absorption properties are not similar between surfaces. Neubauer [46] has also studied more closely the differences in reverberation time predictions using a variety of formulas.

2.2.2 Measurement of reverberation time

For real-world measurements of reverberation time, multiple methods have been developed, each having their own advantages and characteristics. Some of the most commonly used means are presented here.

The most traditional way of analyzing reverberation is to produce whether an impulse-like or truncated sound and measure or listen to the attenuating sound. This is analogous to the definition of reverberation time. While it seems very simple compared to the more advanced methods, it is an easy way to roughly evaluate the reverberation properties. In larger spaces, this method can be used for analyzing the directions of early reflections. With a recorded impulse response, fitting a straight line to the logarithmic sound decay curve is needed for defining RT_{60}. In cases where the signal-to-noise ratio is less than the required 60 dB, or the measurement provides unreliable results, alternatively the attenuation time of 20 or 30 dB can be used when multiplied with 3 or 2, respectively. These measured values are commonly noted with T_{20} and T_{30}. Usually filtered or unfiltered noise, shooting blanks or popping balloons are used to produce repeatable excitation signals. For reliable results, averaged measurements have to be performed in multiple source and receiving positions [33].

Measurement methods using pseudorandom noise sequences, such as maximum length sequence (MLS), require dedicated signal processing equipment in the analysis stage for calculating the correlation between a synchronized source and the receiver. One of the advantages in MLS method is the ability to measure in situations where the signal-to-noise ratio is very low. According to Lahti [33], by using long measurement sequences, SNR can be increased by over 80 dB. Also, the ability to function at low sound pressure levels makes this method suitable for situations where discreet measurement is required. One important weakness in MLS measurement is the dependency on time invariance. As modern active acoustic implementations are often incorporating time-variant properties for controlling acoustic feedback, the MLS method is not usable in these cases.

Logarithmic sine-sweep measurement, however, is an usable method when high sound pressure levels are not problematic. A paper by Müller and Massarani presents a comparison between sweep and MLS techniques. Advantages for the sweep measurements are equally good resistance for background noise, loudspeaker nonlinearities and distortion, but most importantly regarding this work, the ability to reliably measure time-variant systems [44].

Independent of the method used to measure reverberation time, the measurement result can be often used in order to define the average absorption coefficient in a room by measuring the effective dimensions of the space and utilizing suitable reverberation time formula to calculate the absorption. This is a widely used practice especially in sound insulation measurements.

A plotting technique called Schroeder integration is a method of integrating a measured or calculated impulse response starting from the end. An advantage of such a procedure is that the resulting smooth decay curve is easily interpreted even with noise excitation signals [33]. Limiting the integration to the actual impulse response and neglecting the excess noise is necessary for the efficient use of this technique [50].

2.3 Concert hall acoustics

As has been mentioned earlier, the impulse response in a closed space defines how a certain room sounds. The temporal period of early reflections is regarded as one of the most important factors related to hall acoustical descriptors. More accurately, the relation between amplitudes and delays of the direct and reflected sound defines the impression of a hall received by the listener.

Some of the most commonly used descriptors for acoustics especially in concert halls are presented in the following. For a specific area of reverberation, *liveliness* is used to describe the reverberation time at middle and high frequencies, normally above 350 Hz. *Intimacy* is related to the perceived size of the hall, where a large space giving an impression of the performers playing closer to the listener than in reality, is a preferred feature. Also a good *envelopment* gives an impression of being surrounded by the performance, instead of following it from a distance. In addition, there are plenty of other descriptions, such as *blend*, *ensemble* and *spaciousness* [42, 57]. Generally all the terms above require binaural measurements for obtaining numerical results for objective comparison. Since the direction of reflections are considered as the most significant factor in spatial impression, monaural measurements do not provide sufficient information about concert hall acoustics.

The most commonly used properties that can be seen from monaural impulse responses are *early decay time* (EDT) and *clarity* (C80). EDT is similar measure to the reverberation time, but in this case only the deacy time for the first 10 dB is measured. EDT is interpreted by the listener as a clue of reverberation time RT_{60}. Clarity on the other hand is calculated as a fraction of integrals from squared impulse response $h^2(t)$

$$C_{80} = 10 \lg \frac{\int_0^{80ms} h^2(t)dt}{\int_{80ms}^{\infty} h^2(t)dt} \text{ [dB]} . \tag{2.33}$$

In practice the result describes how easily or clearly transients and changes in the sound source are perceived at the receiving point, whereas higher numbers represent more transparent clarity [31]. *Definition* (D50) is similar measure, but the time limits are different [4]:

$$D_{50} = 10 \lg \frac{\int_0^{50ms} h^2(t)dt}{\int_0^{\infty} h^2(t)dt} . \tag{2.34}$$

According to Kuttruff [31], definition describes better speech intelligibility, whereas clarity is more closely related to music acoustics. Barron [4] has stated that for definition parameter, the 80 ms time interval should be normally used for music application, while the 50 ms interval is more suitable for speech.

The effect of a room to the experienced sound pressure level is indicated by a strength parameter *G*. Generally, it is defined as the ratio of the total energy of the impulse response and the measured energy at 10 m from the source [50]. In situations where the acoustic properties are changed, it is also possible to compare the energy from two measurements performed in the same position to obtain the relative change in strength.

$$G_{relative} = 10 \lg \frac{\int_0^{\infty} {h_1}^2(t)dt}{\int_0^{\infty} {h_2}^2(t)dt} \text{ [dB]} . \tag{2.35}$$

Besides the basic reverberation time, the property of *running reverberance* (RR) exists to describe reverberation conditions between the stage and hall. RR defines how well the performers on the stage are supported by the reverberation from the hall. According to Gade [22], the support consists of the perceived reverberation during breaks and pauses that bind the consecutive notes. A descriptive number for running reverberance is calculated by comparing the total sound energy between two 160 ms periods:

$$RR = 10 \lg \frac{\int_0^{160ms} p^2(t)dt}{\int_{160ms}^{320ms} p^2(t)dt} . \tag{2.36}$$

According to Möller, subjective parameters of reverberance and liveliness are mostly connected with the early decay time, neglecting the RR parameter. Like the envelopment and intimacy parameters, also spaciousness and apparent source width (ASW) parameters all depend on the objective binaural measurements of early lateral energy fractions and interaural cross-correlations [42].

In general, reverberation times suitable for different types of music vary in relation to the room volume. For instance, for chamber music in 500 m^3 room an average reverberation time of 1.2 s is recommended by Möller, whereas symphony orchestra needs approximately 2.0 s in 3000 m^3 hall [42]. According to the same reference, desirable values for orchestra rehearsal is about 1.5 s. In fact, requirements for shorter reverberation during practice is rather inconsistent with the effect of audience in concert halls. Since the audience creates additional absorption, the reverberation time is shorter than in empty hall. The relative change in absorption is the greater the more modestly the seats are upholstered. The volume occupied by the attending audience is considered negligible in this case.

2.3.1 Stage acoustics

Although in performance spaces the hall and the stage are inseparably connected, the acoustical conditions have different requirements for both sections. Orchestra pits in operas present yet special needs for the co-operation between the orchestra and singers. However, in this section the discussion is concentrated only on traditional stages found in concert halls.

While the audience usually expects to hear the performed music as whole, balanced and detailed, the performers preferably require additionally such acoustical environment that performing at optimum level is possible, since the acoustical conditions perceived by the musicians have a direct influence on the interpretation and playing style. Most importantly, the players should be able to hear themselves clearly, so that possible corrections in playing technique would be possible in order to adapt to the current acoustical conditions. Thus the selected tempo, volume and articulation depend on how the performer perceives the played music, which is usually different from the music heard in the audience. The space itself should provide support to the performer in a way that the instrument is acoustically connected to the hall, and not just being a distant sound source.

In the case of performance by an ensemble, there is certain measures to describe how well the musicians can hear each other and how the sound of individual performers are blended. As an extreme example, an orchestra, where the players could only hear themselves playing, would definitely not perform with the same quality as in normal circumstances due to the loss of synchronization, balance and the perception of integrity.

According to Nummela [48], in acoustical design, the aspect of the performers has not been taken into account until the end of the 20th century. According to Meyer [41], beginning from the 1980's, a series of publications have studied the acoustical conditions on the stage. Meyer has also stated three degrees of quality for musical communication on stage. At the first degree the volume of produced sound of a single player is related to the intonation and rhythmic precision and at the second degree the quality of sound is considered against the acoustical support. The third degree emphasizes the integration of the whole orchestra to produce balanced and precisely articulated music as discussed above. Also the importance of the acoustic conditions at the conductors position is pointed out as the conductor can be seen between the audience and the orchestra from the acoustical aspect.

As for the audience, early reflections are important as well for the performers, since the earliest response even in small halls is received from the surrounding walls and the ceiling after the floor reflection. Early reflections have to be also well balanced compared to the late reverberation from the hall to give sufficient clarity. Support at the stage is usually presented with separate early, total and late support parameters [23], respectively:

$$ST_1 = ST_{early} = 10\lg \frac{\int_{20ms}^{100ms} h^2(t)dt}{\int_0^{10ms} h^2(t)dt} \tag{2.37}$$

$$ST_2 = ST_{total} = 10\lg \frac{\int_{20ms}^{1000ms} h^2(t)dt}{\int_0^{10ms} h^2(t)dt} \tag{2.38}$$

$$ST_{late} = 10\lg \frac{\int_{100ms}^{\infty} h^2(t)dt}{\int_0^{10ms} h^2(t)dt} \tag{2.39}$$

According to Möller [42], total support parameter has been found to have correlation with the musicians perceived support with his own instrument, while ST_{early} describes more the ability to hear other musicians on stage. Similarly to the hall clarity parameter (equation (2.33)), clarity on stage (CS) is used to describe the musicians feeling of reverberance:

For all the ST parameters discussed above, the measurements are performed by using a omnidirectional microphone at 1 m distance from the sound source [42].

$$CS = 10\lg \frac{\int_0^{80ms} h^2(t)dt}{\int_{80ms}^{\infty} h^2(t)dt}. \tag{2.40}$$

Gade has stated that ST_{late} parameter would replace CS parameter for the measure of reverberance on stage. Besides for the audience section of a hall, the early decay time is also used to describe the impression of timbre or frequency response on the stage. Möller [42] suggests that the approximate value for stage would be 30% lower than the values measured in the audience. Gade [23] on the other hand has presented that evaluating averaging equation of early decay time at selected frequencies

$$EDTF = \frac{EDT(250Hz) + EDT(500Hz)}{EDT(1kHz) + EDT(2kHz)} \tag{2.41}$$

can be used to describe the tonal characteristics or brightness perceived by the musicians on stage.

The shape and dimensions of the stage have been studied by Nakamura *et al.* [45] by gathering subjective opinions from members of various orchestras. Three general types of stages were involved in the investigations, of which conventional-type stages was the most common. Also a small number of sound-diffusing and arena-type stages were studied. In the conventional stages the walls were rather smooth compared to diffusion-type stages that had plenty of sound scattering surfaces on the walls of the orchestra shell. Results of the study suggested that the musicians preference correlated negatively with the increased inclination of the shell walls. This would suggest that with stages featuring a strongly fan-shape, the early reflections are directed away from the orchestra. Also a small size of the stage was seen to have a negative effect. In the study of Nakamura, 1000 m^3 was considered the minimum for an acceptable stage volume.

Barron [4] has suggested an average area of 1.5 m^2 to be reserved for each string and wind instrument and an area from 10 m^2 for large percussion instruments. In reality, the net area covered by a full-scale symphony orchestra of 100 members is approximately 150 m^2. Nummela [48] suggests that in addition to the clearance at the sides of the stage, also an area of 1 m wide should be left free at the front of stage to provide floor reflections to the audience area.

The height of the ceiling is also an important consideration from the aspect of total stage volume. By having a ceiling excessively high, reflections are delayed, which makes the playing in an ensemble more difficult. According to Gade [23], early reflections should not be delayed more than 35 ms from the direct sound in ideal conditions, which results in the maximum distance of 6 m for reflecting surfaces above. In cases where the stage is designed to have a height higher than this, a popular solution for providing early reflections for the orchestra without reducing the physical volume is to suspended reflective elements from the ceiling.

Chapter 3

Electro-acoustics

In addition to the traditional design paradigms for room acoustics, modern audio processing systems can be used to whether enhance or modify the room acoustical properties or to create a totally new acoustical environment. The latter alternative is often used in listening situations to simulate a certain impression of acoustical space. Common techniques involving so called auralization systems are filtering and convolution, but also complex physical models can be used to give the original sound new surroundings [29]. This kind of electro-acoustics is commonly featured in consumers audio products, such as computer sound devices and also utilized in architectural simulations, virtual reality or even in a high-quality car audio systems [3, 13, 15, 24]. However, since auralization techniques are not directly in the scope of this work, the subject is not discussed further here.

Another approach to electro-acoustics is a system being functional inside an existing acoustic space. The functioning of these kinds of solutions depends on real sound fields, and the processing is performed to add or remove certain acoustical characteristics, of which the reverberation properties form clearly the most popular application. From the equipment point of view, besides the signal processing needs of auralization systems, equipment familiar from sound recording and reproduction are needed. Systems having reverberation enhancing properties and featuring receiving and transmitting electro-acoustic transducers in the same acoustic space are called *regenerative* systems. During the last decades, this scheme has been utilized in a number of halls in order to increase the impression of spaciousness or to fix acoustical design flaws.

By using an electro-acoustic system, it is possible to create a space with variable acoustic conditions. This permits a single hall or large room to be suitable for multiple uses, such as classical concerts, theater plays or opera performances. Implementing a system with variable acoustics in an existing hall can be economically much more viable alternative compared to building several separate halls. Practically all electro-acoustic systems installed in music halls have been designed to provide better acoustics for the audience.

3.1 System classification

The acoustical systems shortly described earlier represent different approaches to active acoustics. Kleiner and Svensson [30] have proposed a classification of acoustic systems according to their operating principles. In addition to the active systems, also some passive methods are included in this reference, such as auxiliary reverberation chambers embedded to the actual room. In the following, the basic principles of active acoustic systems are presented.

Of non-reverberant systems, a traditional public address sound reinforcement can be seen as a very basic reference for most of the active systems included in the classification by Kleiner and Svensson. In this scenario, the signal chain consists of a microphone (receiver), an amplifier (processor) and a loudspeaker (source). This chain then works as a single-channel direct sound enhancement system. Sound reinforcement systems often utilize microphones and loudspeakers with at least modest directivity. With this and carefully selected microphone positions close to the original sound source, acoustic feedback problems causing coloration can be usually circumvented by maximizing the direct sound and minimizing the amplified sound arriving to the microphone, which also renders the system non-regenerative. However, if the acoustic feedback occurs, coloration in this case is easily audible usually in very characteristic howling sound. Although such systems can be transformed into reverberant systems with suitable reverberation equipment, this type of systems are neither naturally reverberant nor regenerative, thus for enhancing the real acoustical properties of a room, they are hardly suitable. [30]

Enhancing early reflections in a non-reverberative manner is a less used method in active acoustic systems. According to Kleiner and Svensson, wave front synthesis, employing arrays of loudspeakers and microphones to generate additional early reflections in larger halls, is the most advanced concept in this area [30]. This technique is usable in situations where the shape of the hall does not provide enough early reflections from side walls or ceiling. Commercial concepts using this principle, for instance Acoustic Control System ACS by Berkhout *et al.* [7], have been developed earlier. Nummela has stated that the implementations of early reflection enhancement systems tend to work more or less as regenerative systems in contrary to the wave field synthesis concept due to the natural acoustic feedback occurring in the system [48].

The last of the three active system types is an enhancement for reverberant sound, which can be accomplished by either regenerative or non-regenerative methods. While the non-regenerative option uses the same operating principles as the early reflection enhancement described above, the regenerative alternative is mainly based on the acoustic feedback in the system. The reverberant sound is amplified and then directed to the loudspeakers to create an impression of additional reverberation. While the amplification alone can provide adequate enhancement, the reverberation can be increased with a signal processing algorithm [30].

Instead of classifying active systems purely on the basis whether they are reverberative or regenerative, audio bandwidth affected by the system or the impulse response type can be also used as the classifying factor between active systems. In multi-channel application, a wide-band solution affects to the whole audio frequency range simultaneously in all channels, whereas narrow band systems consist of separate channels for narrow frequency bands each. Hence, a narrow band system generally requires a large amount of channels, the number being reciprocally proportional to the selected bandwidth of a single channel. An advantage in this solution is a comparably good resistance against coloration and instability [48]. A large number of channels is also required for achieving sufficient amplification. In contrary, nowadays the most often used applications utilizing wide-band systems need much less channels, but as a consequence coloration and stability conditions must be carefully examined. Since the room and acoustic system are coupled in series and amplification or additional reverberation is added to the signal, control of regeneration is required in most cases to prevent excess feedback. This issue is reviewed in the next sections.

According to the classification by Svensson [61], the active acoustic system utilized for the investigations in this work is of a regenerative type. As the loudspeakers and microphones are placed in the same acoustical space, the traditional problem is the resulting acoustic feedback. Without any control, this can cause sound coloration and easily distinguishable noise artifacts, namely feedback whistle or howling. Some general methods for feedback control are reviewed in the next sections.

3.2 Regenerative systems and signal processing

Since this work concentrates on creating a natural sounding reverberation enhancement in spaces of various sizes, regenerative reverberation is practically the only viable alternative for implementing such systems. In this section, some details are discussed more thoroughly concerning the signal processing in reverberation enhancement.

Before the digital signal processing era, electro-acoustic systems were utilizing analog devices like plate reverberation and tape delay units, even in large-scale installations. As an example, Ditamore [17], dating back to 1965, presented an acoustic control system for 6000 seat fan-shaped multifunction auditorium for combined reverberation and sound reinforcement implementation. Signal chain in that system was rather similar to generic solutions, but with less regenerative properties, thus resembling wave-field synthesis described earlier in this chapter.

Independent of the signal processing method, a model for simple active system presented in Figure 3.1 is theoretically valid for all the technical alternatives. In this example the system consists of two transducers, which are interconnected in the signal processing. The type of possible connections specifies the state of coherence in the implementation. In the figure, the used notations are as follows. The source signal in the frequency domain is $S(\omega)$. The transfer functions $H_{XY}(\omega)$ represent each a specific function from a single source X to a single receiver Y. $\mathbf{G}_{ML}(\omega)$ denotes gain between each microphone and loudspeaker, and $\mathbf{H}_{LM}(\omega)$ expresses the room-dependent transfer function from loudspeakers to microphones.

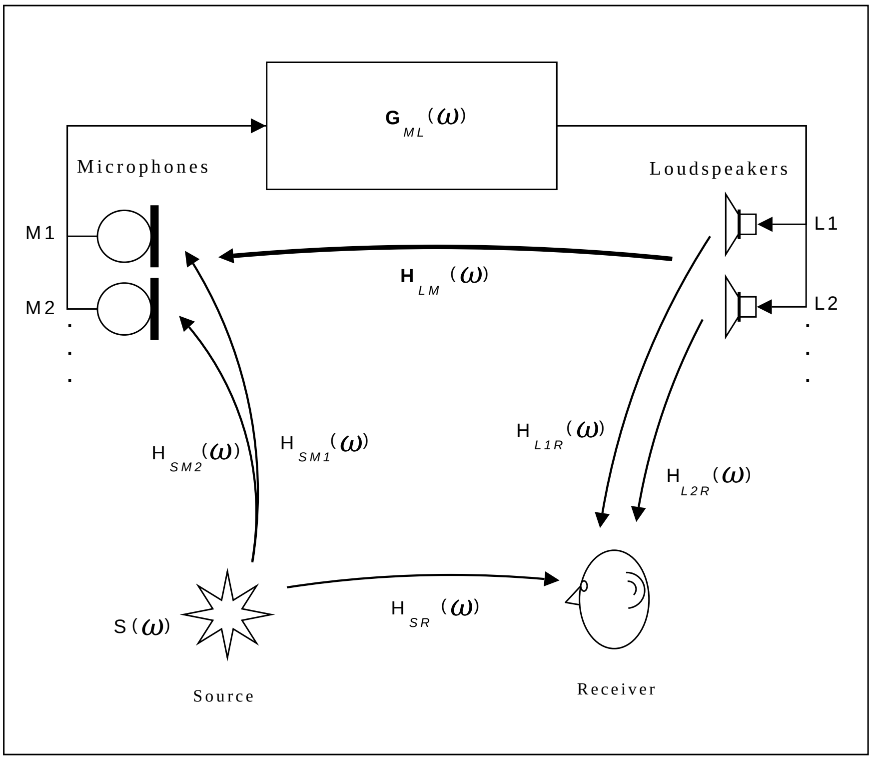


Figure 3.1: A simplified diagram for an active electro-acoustic system.

The transfer function for the signal processing can be seen as a matrix $\mathbf{G}_{ML}(\omega)$ of the size $n_L \times n_M$, where n_M denotes the total number of microphones and n_L the number of loudspeakers. As the coherence of the system is defined by the individuality of system channels, in a non-coherent system, the matrix $\mathbf{G}_{ML}(\omega)$ is diagonal, having values $\mathbf{G}_{ML_{ij}}(\omega) \neq 0$, when $i = j$, $i = 0 \ldots n_M, j = 0 \ldots n_L$. For instance, an unity matrix

$$\mathbf{G}_{ML}(\omega) = \begin{bmatrix} 1 & 0 \\ 0 & 1 \end{bmatrix} \tag{3.1}$$

would represent a system with two separate transducers with ideal frequency-independent unity gain. Griesinger [25] has stated that designing a system with diagonal gain matrix is not sensible due to the excess need of separate channels.

Other transfer functions derived from Figure 3.1 are described next. A column vector $\mathbf{H}_{LR}(\omega)$ of size $n_L \times 1$ includes all the transfer functions from each loudspeaker L_j to the receiver, and correspondingly a column vector $\mathbf{H}_{SM}(\omega)$ of size $n_M \times 1$ includes the transfer functions from the source to each microphone M_i. $\mathbf{H}_{SR}(\omega)$ is the direct transfer from the source to the receiver. Matrix $\mathbf{H}_{LM}(\omega)$ (sized $n_M \times n_L$) expresses the room-dependent transfer function separately from each loudspeakers to every microphone.

For the whole room including the system, the transfer functions from the source to the receiver can be expressed as a sum of the transfer functions of the room H_{SR} and the electro-acoustic system H_{EA}:

$$H_{tot} = H_{SR} + H_{EA} \; . \tag{3.2}$$

The relations for the inputs and outputs of matrix $\mathbf{G}_{ML}(\omega)$ can be formulated from the functions below. Here the notation $S = S(\omega)$ is used.

$$G_{ML11}(SH_{SM1} + Y_1 H_{LM11} + Y_2 H_{LM21} + \ldots + Y_{n_L} H_{LMn_L1}) = Y_1 \tag{3.3}$$

$$G_{ML22}(SH_{SM2} + Y_1 H_{LM12} + Y_2 H_{LM22} + \ldots + Y_{n_L} H_{LMn_L2}) = Y_2 \tag{3.4}$$

$$\vdots \tag{3.5}$$

$$G_{MLn_Mm_L}(SH_{SMn_M} + Y_1 H_{LM1n_M} + Y_2 H_{LM2n_M} + \ldots + Y_{n_L} H_{LMn_Ln_M}) = Y_{n_M} \; . \tag{3.6}$$

The solution of the linear equations is the response of the active system to the receiver:

$$\sum_{j=1}^{n_L} Y_j(\omega) H_{LjR}(\omega) = R(\omega). \tag{3.7}$$

By the definition of transfer function, the response of the electro-acoustic system from the source to the receiver is thus

$$H_{EA} = \frac{R(\omega)}{S(\omega)} \; . \tag{3.8}$$

Additionally, H_{EA} can be expressed in form

$$H_{EA} = \mathbf{H}_{LR} \bullet [(\mathbf{I} - \mathbf{G}_{ML}\mathbf{H}_{LM})^{-1}\mathbf{G}_{ML}\mathbf{H}_{SM}] \; , \tag{3.9}$$

where $\mathbf{I}$ is unity matrix of size $n_M \times n_M$. The complete derivation of the matrix formulation is presented by Svensson [61]. It is noticeable that the gain matrix $\mathbf{G}_{ML}$ appears in both numerator and denominator of the fraction. The denominator part of the equation has a recursive feature, which is directly connected to the feedback level and instability of the system. This is noticed from the product $\mathbf{G}_{ML}\mathbf{H}_{LM}$ that introduces instability to the system, when approaching to unity gain. Colorations can appear at mean gain values much lower than this. In this case, widely studied time-variant properties have not been taken into account in order to prevent the effect [30, 48, 54]. Formal expression for stability condition is called the Nyquist stability criterion [62]:

$$\text{Re}\left[\mathbf{G}_{ML}(\omega)\mathbf{H}_{LM}(\omega)\right] < 1 \;,\; \forall\, \omega \tag{3.10}$$

Poletti has stated that the recursive inverse matrix $(\mathbf{I} - \mathbf{G}_{ML}\mathbf{H}_{LM})^{-1}$ is ”*useful for a detailed understanding of coloration and stability*” but for sound pressure levels and reverberation time gains, mean power analysis would have to be performed. The most straightforward result obtained from this analysis is the gain of overall sound level. The principle is to adjust microphone amplification and loudspeaker amplification to match separately, one channel at a time [54].

In closed spaces, stability and coloration issues are present when audible gains are used. Also, the reverberation algorithm might present possible instability features, depending of the impulse response properties. Howeverm, the loop gain can in normal circumstances be kept low enough to avoid such undesirable effects despite the use of IIR reverberation algorithms.

3.2.1 Reduction of feedback

All the electro-acoustic systems that incorporate acoustic feedback in the signal chain, are prone to uncontrollable feedback or its minor effects of coloration and audible artifacts. Generally, this is caused by a high loop gain, which results from unsuitable amplification settings or poorly placed microphones and loudspeakers in relation to the equipment directivity properties. In generic public address systems, a high amplification combined with careless microphone positioning can often cause issues that in many cases lead to saturating the amplifier to the power output limit at the peak frequency of the system response. In active acoustic systems, especially in small, rectangular rooms, the possibilities for equipment positioning are rather limited, thus the alternative for preventing such problems, besides using inaudible gain levels, is the use of time-variant properties in signal processing. Commercial products exist utilizing spectrum analysis and narrow band-stop filters for preventing feedback in public address systems [6]. The main objective in the time-variant implementations is therefore maximizing the *gain before instability* (GBI) without introducing unwanted artifacts to the sound.

Besides the use of physically moving microphones, the first implementations of time-variant systems by Burkhard [10] and Schroeder [59] utilized analog filters with frequency shifting characteristics. Burkhard approximated a possible 8 dB increase in gain with modulation frequency of 5 Hz. However, this method is not particularly suitable for uses where stability of musical pitch is important due to the fact that the output of such a system is constantly detuned compared to the original sound. Although the subjective testing performed by Burkhard revealed no noticeable artifacts, frequency shifting is mainly used only in speech applications. Recent studies of frequency shifting by Poletti [55] include the discussion and analysis of reverberation enhancement in multi-channel systems.

Nielsen [47] has reviewed more thoroughly the differences between the other time-variant methods in addition to the frequency shifting. Whilst using equalization to flatten the room frequency response with time-invariant feedback systems, modulation techniques generally employed in telecommunication are usable to some extent in time-variant implementations. These modulation methods include phase modulation (PM), frequency modulation (FM) as well as delay modulation (DM). All these modulation methods are of angle modulation type. Modulation signal in time-variant systems is typically a sine wave or other periodical signal, which provides the possibility of controlling the modulation parameters, since in practical cases the input signal, functioning as a carrier wave, cannot be regulated. A common feature for all these methods is that the audio signal (as carrier in this case) is modulated with the modulating signal phase angle $\theta(t)$ of instead of the amplitude, as in amplitude modulation.

While the modulating signal $x(t)$ in telecommunications is the data signal, in a time-variant application the modulating signal is generated [48]. Assuming that the modulation signal is periodic, the filter results in a linear, periodic time-variant filter (LPTV), where modifying the frequency response of $x(t)$ is the method of parametrization. In frequency domain, a LPTV filter causes symmetric sidebands next to the input signal frequencies separated by multiples of the modulation frequency f_m, while a non-periodic filter causes continuous sidebands. Frequency response of such modulation can be presented as following:

$$Y(f) = \sum_{k=-\infty}^{\infty} G_k(f - bf_m)X(f - bf_m), \tag{3.11}$$

where G_k and b denotes the sidebands transfer function and the number of the sideband, respectively [47].

By the basic principles of signal processing [11], modulated signal $y(t)$ is defined

$$y(t) = f(x(t)). \tag{3.12}$$

In complex domain the equation of general angle modulation has the form

$$x(t) = e^{j\omega t} \Rightarrow y(t) = \theta(t)e^{j\omega t} = e^{j[\omega t + \theta(t)]} \tag{3.13}$$

and in frequency domain the equation converts into a convolution [47]

$$y(t) = x(t)e^{j\theta(t)} \overset{DFT}{\leftrightarrow} Y(\omega) = F\{e^{j\theta(t)}\} \otimes X(\omega). \tag{3.14}$$

Here the phase modulation with instantaneous phase angle $\theta(t)$ is used as a modulation signal, where

$$\theta(t) = \beta m_p(t), \tag{3.15}$$

β denoting the maximum phase deviation in radians and $m_p(t)$ denoting the phase modulation function. Sine wave of certain frequency is the simplest case of $m_p(t)$, although other waveforms can be used. Svensson *et al.* [62] have stated that a pseudorandom modulation signal would not result in significantly better stability than with sinusoidal signal.

In the case of FM, the modulation function differs from closely related PM modulation (equation 3.15). The equation for frequency variation is

$$f(t) = \delta_f m_f(t), \tag{3.16}$$

where δ_f and $m_f(t)$ denote the maximum frequency deviation in Hz and frequency modulation function, respectively.

Delay modulation is the third common angle modulation type. The average delay incorporated to the signal is denoted by τ_0, while the modulation function is

$$d(t) = \tau_0 + \delta\tau m_d(t), \tag{3.17}$$

which is analogous to the other modulation functions above.

Frequency shift (FS), used for instance by Schroeder [59], is equal to the PM technique with modulation function

$$\theta(t) = 2\pi f_m t. \tag{3.18}$$

Here the modulation index of phase modulation β equals 2π, thus wrapping the phase. f_m is the maximum frequency shift in Hz [47].

Of the techniques above, according to Nielsen [47], DM is a popular modulation used in commercial reverberation enhancement system due to its rather straightforward implementation. Studies comparing the efficiency and audibility of discussed modulation methods have been performed by Svensson [61], whereas many publications concentrate on discussing only a single technique.

For all modulation techniques in electro-acoustic systems, an equation by Schroeder [60] defines the mean loop gain parameter *MLG*

$$MLG = \overline{|S(f)|^2}, \tag{3.19}$$

where the system response over effective frequencies is averaged. From this equation the definition for *gain before instability* is derived, indicating the performance of electro-acoustic system

$$GBI = 10 \lg MLG \text{ [dB]}. \tag{3.20}$$

The value for GBI depends on the irregularity of the loop gain $S(f)$, where a very uneven gain through the effective frequencies lowers GBI. Schroeder has derived a formula for evaluating expected GBI from the loop gain

$$\langle GBI \rangle = -10 \lg [\lg (B \frac{RT_{60}}{22})] - 3.8 \text{ [dB]}. \tag{3.21}$$

In this equation bandwidth and reverberation time are denoted with B and RT_{60}, respectively. The equation is valid only in diffuse sound fields.

Also, a theoretical maximum value of 2.5 dB for GBI, attainable with FS modulation, is calculated by Nielsen and Svensson [47]. Schroeder, on the other hand, has stated that traditional values for GBI without feedback control are approximately -10 dB. This indicates that considerable improvement can be gained with various feedback control methods in active systems. According to Nummela [48], approximate GBI values of -4 dB and -1 dB have been reached by Nielsen and Schroeder, respectively with time-variant systems.

3.3 Microphones

The type and properties of the microphones used in active acoustic systems have a great effect on how the system sounds. However, the positions where the microphones are placed have an even more important role in the system configuration. Of the two most common microphone types, condenser microphones are typically used in active systems due to the higher sensitivity and linear frequency response compared to those of dynamic counterparts.

Naturally, a frequency response for an optimal microphone should be as flat as possible to capture uncolored sound from the space. Dynamic ranges of over 130 dB featured in professional quality condenser microphones [2] are sufficient for capturing the sound pressure levels produced by the traditional orchestral instruments, especially when not placed in the vicinity of percussion instruments. A high signal-to-noise ratio is also required to maintain the captured signal free from excess background noise.

Directivity requirements for microphones in use are related to the positioning of the microphones and loudspeakers as well as the regenerative features of the system. Omnidirectional and cardioid patterns are most commonly used in all reverberation enhancement systems. In modern microphones, the directional pattern is changeable, thus making it possible to experiment with different settings without having to replace the microphones. Selection of used directivity pattern depends on in what relation the direct and reverberant sound are intended to be captured. With a cardioid pattern directed to the source, the reflected sound is neglected more than in the case of omnidirectional pattern.

Recommendations for the microphone positions are often found in papers discussing reverberation systems generally. Vuichard and Meynial [69] and Lokki *et al.* [38] have presented discussion of using individual microphones for each instrument in an orchestra. While Griesinger [25] has suggested using pickup microphones, the implementations based on per-instrument approach are rather awkward. Since the change in playing position can result in a considerable variation in captured sound pressure levels, close microphones require constant adjustments.

Using a set of microphones in the reverberant field have been a popular method for microphone positioning in earlier studies of regenerative systems. Vuichard and Meynial [69] have stated that the minimum suggested microphone distance from the source is 0.2 times the reverberation radius, where the diffuse field assumption should still be approximately valid. Griesinger [25] has used a microphone pair positioned at the front of the balcony in a theater. This position is a good example of positioning the microphones in the diffuse field. The hall in question seats 1500 people.

3.4 Loudspeakers

The loudspeakers in an active acoustic systems represent the other electro-acoustic interface between the system and room acoustical environment. In addition to the microphones, also the performance of the loudspeakers should exceed the frequency response and noise requirements. Considering orchestral music, fundamental frequencies from traditional instruments range from 40 Hz to approximately 4000 Hz [57]. Although some organ and drum instruments are capable to produce frequencies even lower, the response at high frequencies is more crucial, since the harmonic frequencies are important for the correct timbre of the instruments.

The directivity of the loudspeakers is an important factor. In active acoustic systems, omnidirectional radiation pattern is the most desirable when reaching for an uniform effect in the acoustic space. Unfortunately in most loudspeakers suitable for active systems, the sound directivity is noticeable in high frequencies. These impairments can be circumvented by choosing more distant loudspeaker positions from the listening area, thus extending the sound field of flat frequency response. Also a larger set of equally spaced loudspeakers helps in creating an uniform sound field.

In order to create convincing reverberant room with an electro-acoustic system, the listener must not be able to distinguish the sound coming from a single or multiple loudspeakers. In real acoustic space, the early reflections and late reverberation arrive from nearly infinite number of directions, thus the situation with acoustic system have to be as real as possible. Naturally, the reverberant sound in larger spaces, such as at the concert hall stage, can have a noticeable general direction.

A sufficient loudspeaker distance from the listener in relation to the distance between adjacent loudspeakers is an essential factor in preventing localization effects. Since the human hearing has a limited accuracy of reliably localizing single sources, positioning loudspeakers closely in a row or array formation is an effective way to weaken a sense of separate sound sources if the reverberation algorithm output signals are uncorrelated.

3.5 Existing solutions

While several active acoustic systems has been developed for enhancing the acoustics in performance spaces, only few are designed for practice rooms. In this section, existing electro-acoustic systems are briefly presented.

In Royal Albert Hall, London, a system called Assisted Resonance (AR) was installed in the 1960's to produce additional reverberation to otherwise dry hall by electro-acoustic means. For the system itself, a narrow-band solution consisting of approximately 100 individual channels was used. The AR system in Royal Albert Hall has the effect of providing 0.4-0.5 s longer reverberation time at middle frequencies compared to the same hall without the system [8, 30].

The LARES system [25, 26], originally developed by Griesinger for Lexicon Inc., is a commercialized virtual acoustic product similar to the AR system. This system also is meant to improve the acoustical qualities mainly for the audience, not the performers. The LARES system is possible to be installed for music rehearsal spaces, which on the other hand is closer to the subject of this work. This product under the name Wenger Corporation

V-Room includes a small sound insulated room in which the electro-acoustic system is installed to provide multiple acoustical settings. Systems of this type are in use in a number of locations in United States [34]. Nummela [48] has compared the performance of LARES system and the time-variant reverberation algorithm discussed in Section 4.5.

The CARMEN system, developed in France by CSTB [14], provides an active acoustic solution for performance halls. The product is based on a number of units, consisting of a microphone and a loudspeaker, which are positioned on the walls and ceiling of the hall. With a small number of presets, the acoustics can be modified suitable for several music styles, such as chamber music, symphony orchestra or opera. In Brighton Dome, England, the system provides a reverberation time approximately 1 s longer than without a system, depending on the selected system configuration. CARMEN system is presently installed in nine multipurpose halls in France and England. A new system, called Carmencita, has been recently developed further to function with smaller amount of equipment [58].

Information about other practice room systems besides the V-Room is not widely available as in most of the publications variable concert hall acoustics are discussed. Notable studies include Variable Room Acoustics (VRA) system by Poletti [54], Acoustic Control System (ACS) by Berkhout *et al.* [7] and the Electronic Reflected Energy System (ERES) by Jaffe and Scarbrough [27]. Of these systems, ACS is commercialized by Acoustic Control Systems B.V. [1] and the product is in use in various halls, churches and auditoriums. While the most recent versions provide the possibility for early reflection and orchestra shell acoustics enhancement, the system is intended for performance spaces, not rehearsal rooms.

3.6 Psychoacoustics and hearing

While the domain of psychoacoustics is generally not directly related to electro-acoustics, in this work these two branches are closely connected. Therefore the discussion concerning psychoacoustical perspective is presented here.

For designing and implementing acoustic systems utilizing active components, it is important to understand some of the basic functions of hearing and auditory systems. The main considerations are connected to the sense of sound direction and coloration. Since the purpose of active acoustic systems is to produce artificial environments in spaces that would naturally sound very different and often present contradictions with visual impression contradictions, the acoustic system must be convincing enough to create pleasant acoustical surroundings.

Localization

Accurate sense of the sound direction has been a vital feature in hearing during evolution, which is originally the reason for the development of the spatial hearing. In the case of single sound without additional reflections caused by the room, the localization of sound in auditory systems utilizes certain differences in the sound arriving at both ears. As located on the both sides of the head, ears receive the arriving sound with a slight time difference. As the head is assumed to be spherical with diameter $D \approx 18cm$, for a sound coming from either side of the head, inter-aural time difference (ITD) can be approximated by a formula

$$ITD = \frac{D}{2c}(\varphi + \sin\varphi), \tag{3.22}$$

where φ represents the azimuth angle of incident sound. Maximum ITD value of 0.7 ms for delay is calculated with $\varphi = \pm 90^o$, thus the sound arriving directly from left or right side [28].

The head being an obstacle for the incident sound, inter-aural level difference (ILD) occurs in similar situations when the sound is coming from side of the head [28]. Rossing [57] has stated that at 1000 Hz the sound pressure level at the ear nearer to the source is about 8 dB higher, and at 10 kHz the difference is up to 30 dB. At low frequencies the shadow effect is small due to the dimensions of the head being small compared to the wavelength.

In addition to the phenomena discussed above, also the frequency response of the ear canal depends on the direction of incident sound because of the asymmetrical shape of outer ear and reflections from the body. All the matters above can be combined by measurement into a series of head related transfer functions (HRTF) representing the perception of each ear for sound approaching from certain directions.

The accuracy of perceiving the sound direction in horizontal angles is the most reliable at front of the subject, only having less than 4^o margin of error. At both sides the margin is in magnitude of 10^o, which is still fairly accurate. In terms of loudspeaker positioning 10 degrees means approximately 0.5 m distance between distinguishable loudspeakers at 3 m away from the subject [28].

Precedence effect and masking

Besides determining the arrival of single wavefront, even more important feature in hearing is to integrate the direct sound with the room reflections so that the perceived direction of original sound is not distracted by the directions of the reflections. The phenomenon called *the precedence effect* (also the *Haas* effect) enables the sense of spatial sound instead of hearing numerous single sound sources [9].

There are some limits that have to be met in the precedence effect. The time window in which the reflections are assumed to result from the original source is 35 ms, the successive sounds are required to have similar spectra and time envelopes to the original sound and the amplitude in reflections must not be considerably louder [57]. The latter condition is practically never a problem in active acoustic systems with reasonable gain. According to Karjalainen [28], the sound pressure level of the reflections have to be over 15 dB higher than the direct sound in order to prohibit the direct sound to be heard. If single reflections arrive later than the mentioned 35 ms, they are heard as separate echoes from their own directions, not blending to the source. Möller [42] has presented that reflections arriving too early are perceived as sound coloration rather than spatial effect. In a situation where the reflections are received 45^o from the side, coloration effect is noticed in reflections with delays less than 20 ms. Also, with suitable reflection levels the spatial impression could be attained with reflections as late as 45 ms. Very early (approximately 5 ms) reflections at relative sound pressure levels of approximately -10 dB cause false deductions of the original sound source direction. Sound coloration effects due to artificially amplified sound are discussed more thoroughly in Section 4.4.

Chapter 4

Concept model

In this work, a concept model of an electro-acoustic system for practice rooms of various sizes is implemented and studied. In order to respond to the growing needs for practice spaces in music institutions and orchestras, an active system presents a technique enabling the musicians to rehearse in acoustical environment which resembles the real venue of performance. By the means of traditional acoustic design and passive elements, the same objective would require building a copy of the performance space. Naturally, this is rarely possible due to economical limitations. The electro-acoustic system makes it possible to decrease the size of the hall or room but still have sufficient reverberation to support playing solo or in an ensemble. On the other hand, creating a small space with adequate reverberation results in an excessive sound pressure levels. An additional advantage of an electro-acoustic system is that in similarly sized rooms the sound pressure levels can be lowered considerably by artificially creating the needed reverberation.

4.1 Truncated hall concept

The increasing usage of the facilities in many concert and multi-purpose halls decreases the possibilities of the orchestras to rehearse in their principal performance space. This problem has been tried to partially solve by converting rooms from less demanding use to rehearsal halls. With slight room acoustical modifications, the room can provide a space for orchestral rehearsal. However, such solution rarely results in acceptable acoustic conditions that satisfy the needs of the musicians. In this kind of solution common acoustical flaws like focusing and echoes, poor support and reverberation or excess sound pressure levels can be present, due to the fact that the room does not feature proper acoustical design in the first place [32]. In addition, the acoustics in such rehearsal hall do not correspond to the acoustics in the real concert hall.

A concept of truncated hall for orchestral rehearsal has been presented by Nummela [48]. The proposed implementation comprehends a stage section of a concert hall combined with an active acoustic system. Ideally, the stage would be an exact copy of the performance space, truncated at the second row of seats, where an anechoic wall is installed. This wall is equipped with active acoustic system which is used to recreate the late reverberation that would normally come from the audience area of the hall. In the following, the design principles of truncated hall are discussed more thoroughly.

The original concept of truncated hall by Nummela suggests that the active acoustic system is installed at the anechoic wall. An array of loudspeakers and a set of microphones are positioned close to this wall to provide a sound field for the late reverberation created with signal processing. In the optimal case, where the stage is identical to the stage in the real hall, early reflections are also identical on both stages [48]. This requires that the part of the hall not included in the truncated version does not provide any early reflections. The microphone positions are selected to cover a comprehensive area of the anechoic wall to capture the reflections from the stage to the direction of the audience.

Nummela has noted the difference in Schroeder frequencies between the real hall and the truncated hall. Without the active system, the modal density in the truncated hall is considerably lower at low frequencies due to the decreased volume and highly absorptive wall. By producing diffuse reverberation with the system, the modal density on the stage is statistically closer to the properties of a real hall.

The obvious advantage in this setup is that the generation of early reflections can be neglected because the sound behavior at the stage structures creates the characteristic sound field. This is beneficial, since the accurate replication of early reflections in a totally anechoic room and with large number of loudspeakers is rather difficult. Instead, creating diffuse late reverberation is far less challenging task with modern reverberation algorithms.

4.2 Symphony orchestra rehearsal room

From the truncated hall concept model discussed above, a slightly altered model is discussed here to be used in real world testing in a case study. Originally, the truncated hall concept was built into a large recording studio with highly diffuse acoustic properties. As discussed in previous section, one wall was covered with broadband absorptive curtains to provide a non-reflective surface. The equipment of active acoustic system were installed along the anechoic wall.

This prototype can be further developed, so that the virtual acoustics are expanded on the remaining sides of the room as well. The advantage here is that the balance between virtual audience part and the stage is more controllable than in a situation where the acoustics on the stage are predetermined by the acoustical design.

Requirements for the existing room in case of such implementation are somewhat different from the original concept. The recording studio used by Nummela provided very diffuse reflections, which for the side walls of the stage is rather less wanted feature for natural sounding result [48]. On the other hand, while the orchestra shell on a real stage provides realistic early reflections, the natural high reflectivity is probably excessive in order to be enhanced with an electro-acoustic system without acoustical contradictions. Thus, in this case a stage with some reflective properties would be ideal. While the natural reverberation time should be comparably low, some theater venues could feature suitable acoustics for this kind of implementation.

4.3 Damped small practice room

Using small spaces with electro-acoustic system has many common considerations with large rooms. However, shorter distances between the equipment may introduce some challenges that otherwise would not be encountered.

The basic principle in electro-acoustically enhanced small practice rooms is similar to the system in larger spaces. The room should feature reflective surfaces to some extent for producing natural early reflections. However, depending on the natural reverberation time of the room, some absorption is most likely needed to accommodate the artificial reverberation. Otherwise the decay rates and timbre could appear to be too different between the room and the system. Also, too apparent natural reverberation would cause mixed perception with the electro-acoustic system when using clearly distinguishable reverberation parameters. Because of the wide variety of acoustical conditions in existing practice rooms, using such a system in acoustically poorly designed rooms can cause issues that could be addressed as acoustical incompatibility.

The aim for using virtual acoustics in small rehearsal rooms is to provide an environment that is perceived acoustically more spacious and enables to practice in conditions closer to the performance situation. The electro-acoustic system can easily be implemented to have multiple settings for different acoustics. This gives the user more flexibility than merely having only the option to enable or disable the system. According to the quality requirements by Meyer [41], music ensembles could also benefit from the enhanced reverberation in a better integration between the musicians.

4.4 Challenges

Planning and implementing an active-acoustic system is not necessarily a straightforward procedure. This of course depends on the properties of the room where the system is intended to be set up. The main challenge in this work is to decrease the sound pressure levels in practice rooms without compromising the support of the room for playing. The issues encountered in the small practice rooms are different than those of in larger halls, for instance in equipment positioning, feasible reverberation parameters and computing efficiency and delay. In the following, the most important general considerations are analyzed and discussed.

Room damping

Especially in small rooms, sufficient damping is required for creating more natural sounding environment with an active system. Practice rooms that were inspected prior to the case study testing periods were found to have some absorptive panels installed on the walls and ceiling [39]. To loosely follow the concept of truncated hall, compared to basic acoustical treatment, more absorption was needed.

Nummela has used thick suspended curtains in the truncated hall prototype [48]. This method provides easily removable absorption for large areas. Depending on the quality of the curtain material, the absorption is effective at rather wide frequency band (see Table 2.1). Besides the additional damping, some reflective surfaces are required with regenerative systems. Therefore the walls must not be completely covered with absorptive material in order to preserve some of the naturally occurring early reflections and liveliness. Thus, the amount of absorption has to be balanced so that the lack of it does not interfere with the artificial late reverberation. With excess absorption, the regenerative properties of the active system begins to suffer, which results in increasing the system gain and reverberation time for obtaining audible effect.

In small rooms with an ordinary height of approximately 2.8 m, the first reflections from the ceiling give an impression of certain height, which has an undesirable effect on creating an impression of large spaces. For avoiding the issue, damping the ceiling would be required. However, this is often very obtrusive if dark clothes, such as theater curtains, are used.

With active acoustic systems, the ideal absorption material would need reactive properties, thus having a higher absorption coefficient for higher incident sound energy. Unfortunately, such materials do not currently exist. This would enable to have smooth sound decay instead of a rapidly attenuating sound followed by a quieter artificial reverberation.

Acoustic feedback

The acoustic feedback occurring in active acoustic systems is an issue that needs very careful equipment positioning and feedback controlling integrated in the system. The methods of using time-variance, frequency response equalization, controlling directivity or bringing the microphone closer to the source all aim to increase the gain before instability. Unsuccessful system design and installation is perceived in coloration, artifacts or feedback howling. As described in Section 3.2.1, time-variant system design allows much higher GBI before effects caused by feedback are noticeable. However, being closely related to the above discussion of absorption, the problem with feedback in the system can be formulated as the balance between the acoustics of a room and the active acoustic system as well as with the equipment positioning.

With higher signal amplification, the acoustical effect is more pronounced, but as a result the danger of instability increases. This can be prevented by bringing the microphones closer to the source, which leads to losing the effect of natural properties of the room. Introducing more absorption can also reduce the loop gain, but again, the characteristic acoustics will suffer. The advantage of using time-variant components in the reverberation algorithm gives more latitude in finding suitable combination for these variables.

Coloration

The term timbre in music and acoustics refers to the perceived differences in nuances of a sound produced by a certain source. As of the color of sound, it is determined by the shape of the frequency spectrum, comparably to the visual perception of colors. Coloration in acoustics can be established by an emphasized or attenuated frequencies in the spectrum. A typical cause for coloration in traditional room acoustics is reflections that act as a comb filter, attenuating and amplifying frequencies of certain intervals [48, 53].

While coloration in active acoustic systems can be a result of approaching instability at some frequencies, inherent coloration of the artificial reverberation is less apparent phenomenon than effects directly caused by instability. Experienced subjects can easily distinguish even a slight coloration or unnatural feature which reduces the perceived quality of the system thus lowering the acceptance of the system.

Especially in systems of regenerative type a non-flat frequency response of room transfer function is a significant source of coloration. In the frequency response, room modes cause peaks that are emphasized as the room is acoustically coupled with the reverberation system. This leads to longer decay times at the peak frequencies. Other common type of coloration is a metallic sounding ringing, which is induced by a series of harmonic frequencies from a comb filter characteristics in reverberation system [48]. As mentioned in Section 3.2.1, Väänänen has stated that in active systems, metallic artifacts are caused by the lack of modal density [67].

Also a time-dependent periodicity in the reverberation is an effect comparable to coloration. The reverberation algorithm featuring time-variant comb-allpass filters (Figure 4.1) introduces flutter to the generated reverberation with higher modulation parameters [36, 68]. As a downside, a sound from an instrument fluctuating at a very low frequency can be interfered by an active system that fluctuates at a slightly different frequency.

Reverberation modeling

Enhancing the reverberation and impression of acoustic spaciousness for music practice applications can be approached from various directions. In a small practice room, the intended benefit from such installation is to provide a possibility of experiencing the playing as what it would sound like in a larger space. In this case the aim is not necessarily the acoustics of a concert hall stage, but a less confined room. This raises a question whether the acoustics should be similar to a certain existing room.

For less experienced users, the acoustics with a resemblance of a generic chamber music hall could provide a fairly good representation of how the music would sound in a space larger than an ordinary classroom. On the other hand, creating a convincing simulation of a large concert hall in a very small room is difficult due to the limited capability to modeling early reflections.

The situation in larger spaces is rather different in this matter. The dimensions in a room of a normal size stage implies that the first reflections from the walls represent roughly the reflections of a real stage. Therefore adding only the late reverberation of a concert hall could provide a more convincing result. In this case, tuning the stage acoustical parameters to conform the parameters measured from an existing hall is possible to some extent. As an audience part is not included in the rehearsal hall concept, only the parameters measured on the stage have to be taken into account. In an orchestra rehearsal room the primary concern are the same measured values as on an ordinary stage of a hall, such as support, EDTP, RT_{60} and clarity. Most halls have been measured for the standard parameters, so the desirable target parameters are available. During the tuning process, the objective measurements must not be used as the only design criteria, but subjective impression of the acoustics should be evaluated as well.

Processing delay

With a reverberation algorithm running in a generic PC hardware, the required calculation power must be considered. This is a concern mainly in small rooms where an instant response from the system is more important due to the proximity of the loudspeakers. In larger spaces the impact of a larger buffer size, thus longer delay, is less problematic.

A recommended reverberation time by Teuber and Voelker [65] for practice rooms is low, 0.2-0.6 s. Such dry acoustics do not provide much masking that would allow longer delays in late reverberation, in which case the added reverberation from an electro-acoustic system have to begin in the time window of time masking of approximately 20 ms [28]. Longer delays are audible as echos and are easily detected. Griesinger has suggested that the simulated reflected energy should not be delayed more than 15 ms from the direct energy [35].

Suitability for purpose

Regarding the investigations performed in the case studies, an important aspect is to determine what are the acoustic requirements and recommendations for small practice rooms. For instance excess reverberation in small rooms results in unclear sound. This reduces the quality of the practice by masking the playing faults.

Teuber and Voelker [65] have studied comprehensively the needs in rooms for different practice usage. For small spaces of 30-200 m^3, the average reverberation time is suggested to be between 0.2-0.6 s. A practice room of 30 m^3 is not uncommon in music institutes, and the cited 0.2 s reverberation time is very short. However, they have cited that "with the lack of any sound reflection, the impression of the instruments is unbalanced".

In the same study, Teuber and Voelker have not mentioned the use of active acoustics. From this perspective, in the smallest rooms there is not much need for artificial reverberation, since in most cases the reverberation time of around 0.2 s requires the damping to be rather effective. The difference here is that a modest increase in the reverberation does not compromise the clarity. Instead, the intention is to improve the impression of spaciousness. In practice use, an active acoustic solution with selectable settings can provide an alternative for the acoustics in the room. Naturally, it is up to the user whether the selected acoustics is favorable for practice.

For larger orchestra rehearsal rooms, Teuber and Voelker suggest that the difference of acoustics between the rehearsal space and the concert hall should be as small as possible. By using an electro-acoustic system, the reverberation time and timbre in the hall is possible to be roughly simulated, depending on the capabilities of the system. Here, the suggested reverberation time for small 1000 m^3 rehersal hall is 1.3 s. Such a reverberation time can be considered very low for orchestral music, but longer times would result correspondingly in high sound pressure levels with passive acoustics [65].

User interface

The design of an user interface for reverberation enhancement system is not essentially a challenge, but the subject is important to consider. To be used by teachers and music students of various ages, the user interface must be very simple and self-explanatory so that operating the system is straightforward. A limited set of pre-configured options must be programmed, since the subjects cannot be presumed to be able to modify the reverberation parameters manually. Depending on the number of different acoustical settings, switching between presets or disabling the system must be incorporated to the control interface.

Of the commercial products, V-room by Wenger Corp. [70] uses a MIDI interface for communicating with the reverberation control. This solution enables to control the system by using a wide selection of different user interfaces.

PureData environment allows to program simple user interfaces utilizing radio button type selections for switching between the operation modes. Each button having a short descriptive label, controlling the reverberation system with a mouse is not challenging. The actual implementation of the user interface is discussed in the next chapter.

4.5 Reverberation algorithm

In this section, an advanced method for increasing stability in active systems is discussed. For reverberation enhancement, a simple solution is to add a multi-channel reverberator in order to implement a time-variant active acoustic system. Väänänen *et al.* [68] have presented a time-invariant parametric reverberator. This algorithm consists of a number of parallel channels, each featuring a delay line, an IIR lowpass filter and a comb-allpass filter. Each individual filter channel is fed back to itself and then added to the combined input for all channels. According to the preceding studies, this algorithm produces natural sounding diffuse reverberation, well suitable for reverberation enhancement applications [36].

Figure 4.1 [37] shows the structure of the reverberator, which in this case contains 8 channels. Each fed back channel contains a delay line DL_i, a lowpass filter $L_i(z)$ and a comb-allpass filter $A_i(z)$. β represents the reverberation feedback coefficient. The function of the lowpass filter is to simulate the air absorption and the generally higher absorption coefficient at high frequencies.

The beauty of this algorithm lies in a time-variant version by Lokki and Hiipakka [36]. The time-variance is implemented to the allpass filter feedback and feed-forward parameter a_i. This filter is depicted in the lower part of Figure 4.1. Corresponding to the digital filtering principles, the parameter affects to the distance of poles and zeros from the unit circle on the pole-zero plot while the delay z^{-Di} length in samples D dictates the number of pole-zero pairs [16]. By modulating the parameter a_i with a continuous signal, the amplification

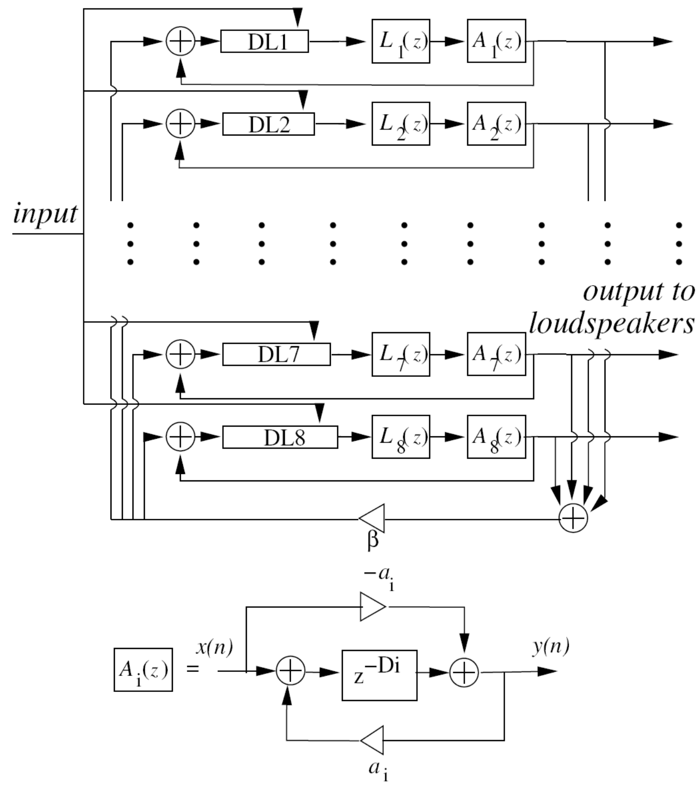


Figure 4.1: A diagram of the time-variant reverberation algorithm.

peaks move slightly in the magnitude response. By carefully choosing suitable values for parameters, the subtle change in the comb-allpass filter response prevents uncontrollable acoustic feedback. The movement of the amplification peaks is shown in Figure 4.2.

A closer analysis of the time-variant comb-allpass filter in Figure 4.1 gives a transfer function

$$A_i(z) = \frac{-a_i + z^{-Di}}{1-(a_i z^{-Di})}, \tag{4.1}$$

from which the pole locations can be derived. By replacing Di with N, pole locations are

$$p_n = \begin{cases} Re^{2n\pi/N}, \ n = 0, \dots N-1, a_i < 0 \\ Re^{(2n+1)\pi/N}, \ n = 0, \dots N-1, a_i > 0 \end{cases}, \tag{4.2}$$

where R represents the distance from the centre of the unit circle

$$R = \sqrt[N]{|a_i|}. \tag{4.3}$$

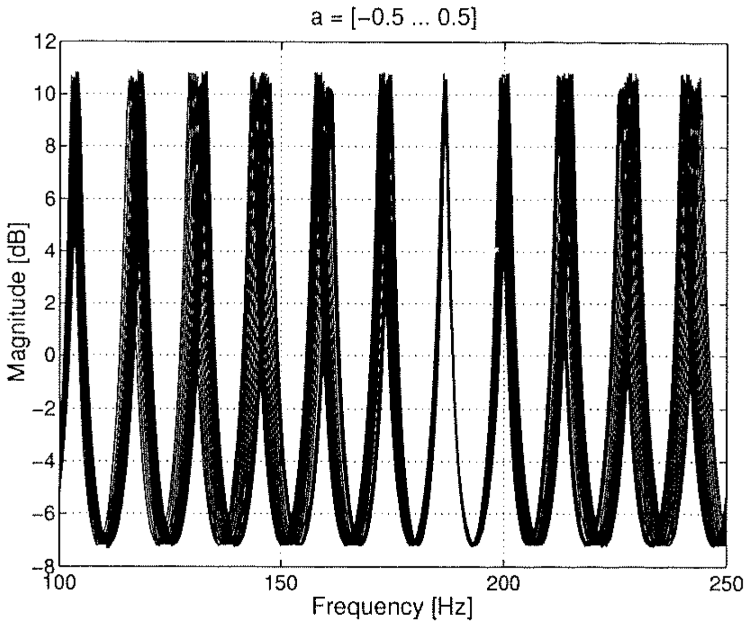


Figure 4.2: A section of the magnitude response of the time-variant algorithm.

Suggested delay line DL_n length values used in an implementation study by Lokki *et al.* [36] were between 67 and 93 ms, corresponding to approximately 2900 and 4100 samples at sampling rate of 44100 Hz. The comb-allpass filter delay lines used lengths from 5 to 10 ms, corresponding similarly to approximately 230 and 440 samples. In the same study a sinusoidal modulation function at 2.5-3.5 Hz were utilized.

Väänänen has stated that using a comb filter may result in a metallic colorated sound effect due to the relatively great absence of modal density in the frequency response [67]. This claim is easily perceivable from Figure 4.2 [36]. However, if the different channels have unsynchronized modulation signals, theoretically the frequency response should remain more flat averaged over a longer time period. Additionally, the comb filters are different in each channel, which reduces even more the possibility of coloration.

The algorithm described above was used in the case studies. Running on Windows platform in PureData environment, a latency of approximately 17.5 ms have been measured earlier [37]. The shortest attainable delay in practice is determined by the available processing power.

Chapter 5

Case studies

The case studies of the active acoustic system form a major part of this work. The tests were performed at the Espoo Cultural Centre in three different spaces in order to investigate the possibilities of such a system to enhance the acoustical qualities in practice and rehearsal rooms. Out of the three rooms, two were in regular use of Espoo Music institute as practice rooms while one case study were performed in a small multipurpose theater called Louhi hall. This venue was prepared to act as rehearsal space for a symphony orchestra of approximately 40 musicians.

In the small practice room cases, the subjects were asked to fill a short questionnaire for collecting their impressions and opinions of the reverberation-enhanced room. The testing period was conducted while these classrooms followed the regular allocation. Both rooms featured the electro-acoustic system for three days. In the Louhi hall case, opinions were collected from musicians with slightly different questionnaire. In this chapter, the spaces used in the case studies are described comprehensively regarding the installation of the electro-acoustic system.

5.1 Cage

The studied first practice room, *Cage*, is located in a bomb shelter area underground. Two adjacent rooms were available, the other being used as a practice room for percussion instrument teaching. Due to the intentions in this work to gather comprehensive comments from scheduled users during the testing period, brass instrument class Cage was selected. The plan of Cage is seen in Figure 5.1. The dimensions of Cage are $5.2 \times 7.2 \times 2.75$ m (width$\times$length$\times$height), resulting in a total volume of 103 m^3.

According to the original design, the room is not intended for music rehearsal at all but it has been later used as such due to the lack of specific practice rooms. Building materials

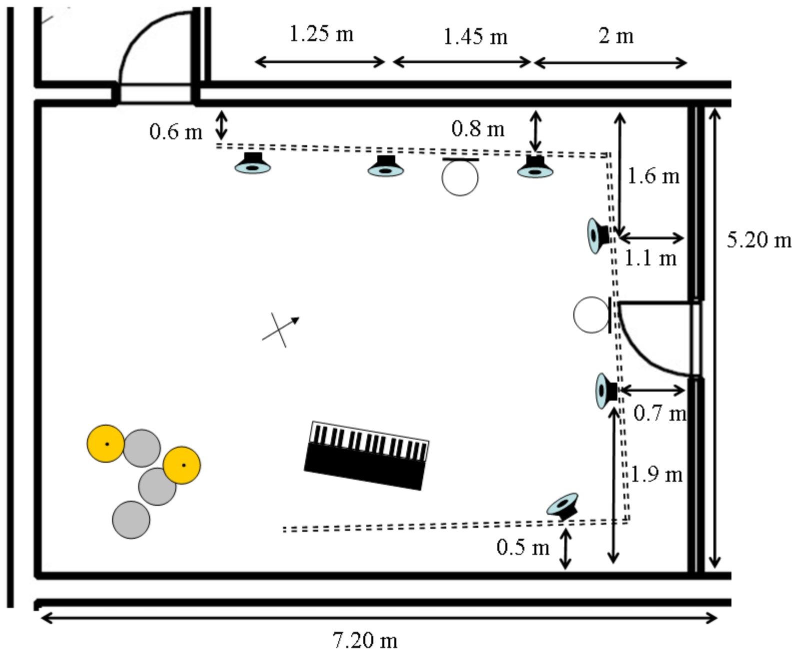


Figure 5.1: Cage, plan and positions of equipment and absorption materials.

consisting of a painted concrete floor, an element ceiling and gypsum board walls are fairly typical for this kind of a space. Besides of a small number of instruments and accessories typical for music room such as an upright piano, a drum set, music stands and tables, the room features very little diffusing objects on the flat walls. Combined to a limited amount of porous absorption material attached directly to the ceiling, standing waves leading to sound colorization are easily perceivable. Actual flutter, however, is not present in clearly audible manner. Lighting and ventilation assemblies are suspended from the ceiling, thus functioning as some diffusing elements. After all, the characteristic acoustics in this room is clearly perceivable.

In the investigations of the practice rooms, background noise level of $L'_A = 31\ dB$ was measured [39]. The noise was caused mainly by the ventilation system. Due to the nature of the room, any particular acoustic design is not expected to have been taken into account in the building and planning stage. Exact impact noise insulation ratings are not available but derived from requirements for structural strength and materials, impact noises such as footsteps are clearly audible from the floor above.

Installation

To reduce the reverberation of the room, thick Molton-type theater curtains were suspended on the top of heavy-duty lighting stands and bars. The curtains, having total width of 10.2 m, formed two incident walls with comparably low reflectivity. The height of the curtain was 2.1 m. Approximately 70 cm of free space was left behind the curtains all the way to enhance the absorptive effect and to accommodate necessary computer hardware.

Equipment for the electro-acoustic system consisted of 6 Genelec 1029A loudspeakers, 2 AKG C414 condenser microphones and a standard PC computer equipped with high-quality RME digital audio interface. For the AD/DA purposes between transducers and the signal processing, an 8-channel external converter was used. An AKG C414 microphone features a selectable directional pattern, a level attenuator and a low-pass filter at 75 or 150 Hz, thus it was a suitable model for the installation [2]. The loudspeakers were positioned along the suspended curtain on approximately equal intervals, while the microphones were located at the middle of both absorbed wall sections at a height of 1.75 m. More accurate equipment positions are shown in Figure 5.1. A general view of the room Cage is shown in Figure 5.2. A system microphone and a pair of loudspeakers can be seen in front of the curtain.

Figure 5.2: Practice room Cage with the system installed.

The reverberation system in Cage featured four presets for reverberation time and strength in the mouse controlled user interface. Each setting was selectable by radio buttons displayed on a monitor and the parameters for each preset are found in Table 5.1. All other

Table 5.1: Reverberation system parameters in Cage.

Setting label	RT_{60}	output gain (linear)
Small hall, quiet (setting *on1*)	1.5 s	0.4
Small hall, loud (setting *on2*)	1.5 s	0.8
Large hall, quiet (setting *on3*)	2.4 s	0.4
Large hall, loud (setting *on4*)	2.4 s	0.8

variables in the reverberation algorithm were fixed during the tests.

5.2 Ives

The second practice room, *Ives*, is located in the third floor of the Espoo Cultural Centre and represents typical practice conditions in Espoo Music Institute. Exact dimensions for the room are 3.8 $\times$ 3.75 $\times$ 2.8 m (width$\times$length$\times$height). The volume of 40 m^3 is very small compared to the official recommendations[1] to provide sufficient room for music practice. As in the usual utilization, accompanying rehearsals are scheduled for this room, which suggests that the regulations are exceeded easily.

However, the room itself is constructed of more suitable materials for music practicing than Cage. With wooden floor, gypsum board walls and some absorption panels, subjective opinion on the acoustics is more neutral than in Cage. Narrow windows are situated at the rear wall. A Steinway model B grand piano and a locker are also found in the room. The plan of the room Ives is shown in Figure 5.3.

Background noise of $L'_A = 33$ dB was measured. The measurement report shows that the airborne sound insulation values R'_w from room to room in this floor are between 52-58 dB [39].

Installation

The system used in Ives is very similar to the one in the first case study room, Cage. The additional absorption was created with the same curtains as in Cage, although the more limited floor space required the use of more compact suspension technique. For this purpose, lighting support tripods were utilized. Of the total wall width of 15.1 m, 11.5 m were covered with the absorbing material. The curtains reached the height of 2.2 m, and the air gaps between the curtains and the walls were in the range of 30-50 cm.

[1]Finnish Institute of Occupational Health regulation 85/2006 recommends minimum volumes of 80 m^3 for a grand piano or a drum set, 30 m^3 for wind instruments and 10 m^3 for other instruments each.

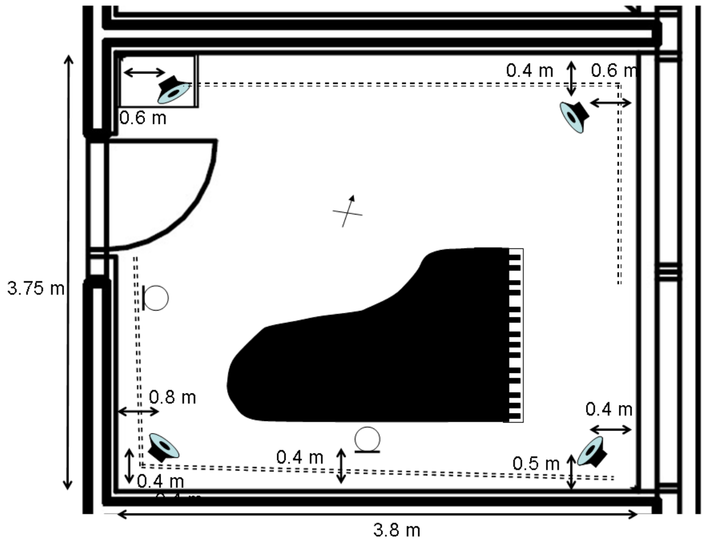


Figure 5.3: Ives, plan and positions of equipment and absorption materials.

User selectable acoustic settings in the system were the same as in Cage, the only exception being the large hall settings (*on2* and *on4*), in which the reverberation time parameter was changed from 2.4 s to 2.0 s. Due to the considerably small floor area compared to Cage, only four loudspeakers were installed, one for each corner of the room. The positions of the equipment used in Ives are depicted in Figure 5.3. A general view of the room Ives is shown in Figure 5.4.

5.3 Louhi hall

Louhi hall is a multipurpose performance space seating approximately 300 people depending on the stage configuration. The hall is primarily designed to accommodate music theater and dance performances as well as small scale popular music concerts with amplified sound. In any case, the hall is not intended for classical music purposes.

The shape of the hall is rectangular, while the audience part features rather steep, constant rise. Balconies surround both sides of the hall to provide space for additional seats and lighting equipment. Also the back of the stage is located under the balcony. The balcony is on the same level with the back row of seats.

The surface materials provide absorption typical to theater spaces. The walls around the stage are basically absorbing panels upholstered with a dark cloth. Some of the side panels can also be turned, while the reverse side provides more reflective surface. Since the side

Figure 5.4: Practice room Ives with the system installed.

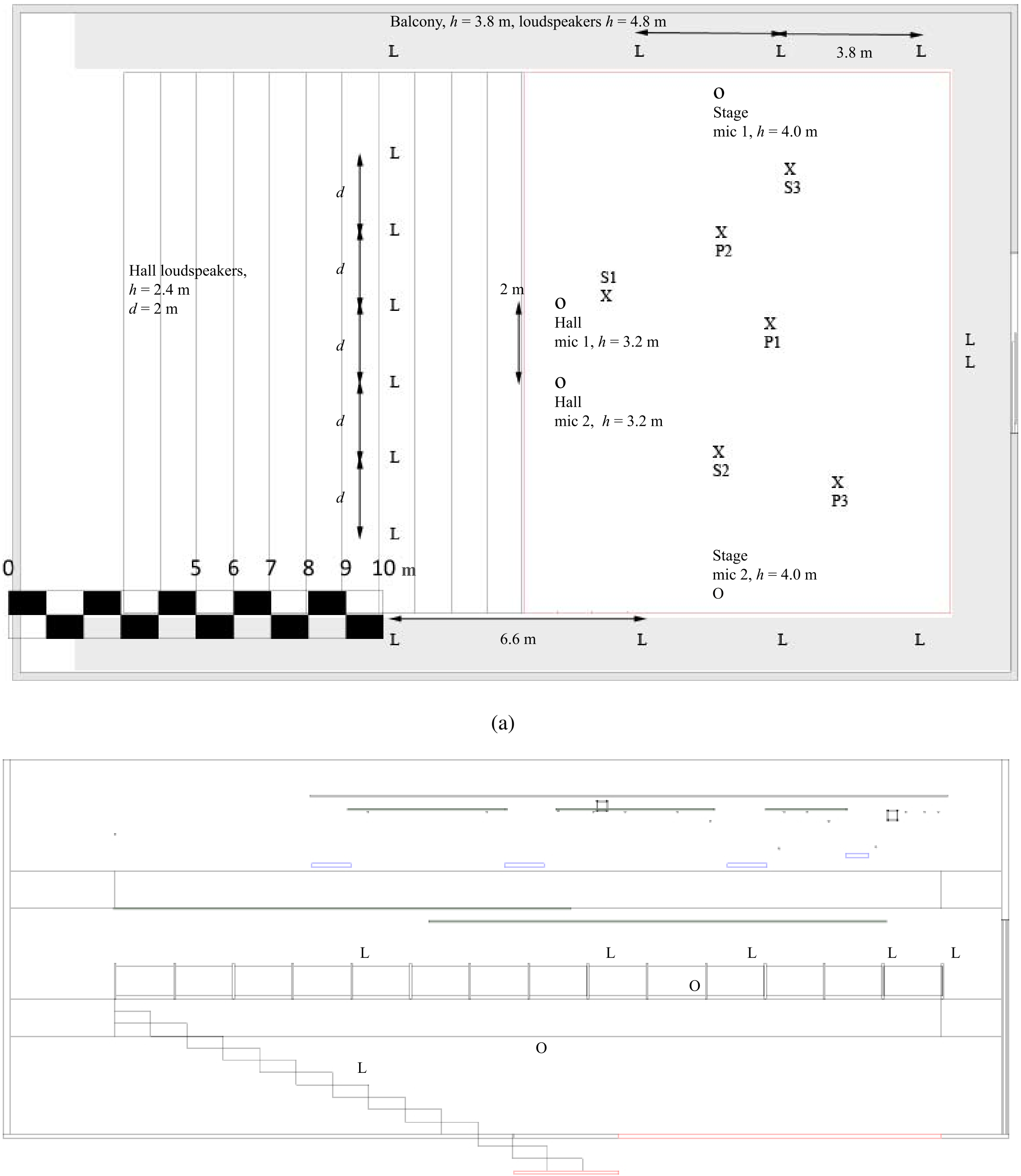
walls are parallel, revealing the reflective side of all the panels causes audible flutter on the stage, thus only two of them were left open in this case.

The ceiling high above the stage is partially covered with absorptive wool. Numerous rails for lighting, suspensions and crane lifter purposes are present with the appropriate light equipment. Multiple layers of light installation were noticed to improve sound diffusing properties, since on the stage, diffusing shapes normally found in concert hall stages were not featured.

The overall dimensions for Louhi hall are 26.4×17.6×9.0 m (length×width×height). The total stage width of 17.6 m is roughly in range with narrow stages found in old shoebox-shaped concert halls, such as Musikvereinsaal in Vienna, Austria. However, the accommodation had to be arranged in this case only for 40 musicians instead of 100. Narrow halls are generally found to provide pleasing acoustics [21, 43]. The depth of the stage is 11.4 m measured from the first row of seats to the front of the balcony. The plan of Louhi hall and the stage can be seen in Figure 5.5(a), where the balcony area is indicated with a darker color.

Installation and tuning

The active acoustic system installed to Louhi hall was an extended setup of the system used in the small practice rooms. This system was based on two standard PC computers on Windows platform. The reasons for using two redundant computers were the guaranteed



(a)

(b)

Figure 5.5: Plan (a) and section (b) of Louhi hall with equipment positions.

processing power and the possibility of dedicating separate systems for running one system for the stage acoustics and the other for the hall simulation. Extremely long cable runs were also avoided by having separate systems for the hall and the stage. A total of 16 two-way Genelec 1029A loudspeakers and four AKG C414 condenser microphones were used in this installation, of which a half were for the stage purposes and the other half for the audience area.

On both sides of the stage, three loudspeakers on discrete stands were installed on the balcony to simulate the energy reflected from the orchestra shell. Due to the considerable directivity at higher frequencies, the loudspeakers were tilted downwards to point more directly to the center of the stage and for delivering more brightness to the sound. The remaining loudspeaker pair were positioned on the balcony behind the orchestra, facing to the sides of the stage. All loudspeakers on the balcony were at the total height of 4.5 m from the stage floor. The positions of the stage microphones were at the balcony floor level at the end of microphone stands that extended approximately 1 m toward the center of the stage.

Six loudspeakers for the hall acoustics were positioned on the fifth row in the audience area at 2 m intervals. These loudspeakers were facing straight forward at a total height of 2.4 m from the stage floor level. The remaining stage loudspeaker pair was positioned on the sides of the audience area on the balconies, facing to the center stage without tilting. Microphones for the hall simulation were positioned at the front of the first seat row, behind the conductor's podium at a height of 3.2 m with a mutual distance of 2.2 m. The concertmaster playing position was 2.4 m in front of the microphones. The positioning of hall microphones was similar to the presented prototype by Nummela [48]. Complete representation of equipment positions in Louhi hall is seen in plan and section Figures 5.5. In these figures, the loudspeakers are marked with letter 'L' and the microphones with letter 'O'. A cardioid directional pattern were used in all system microphones. On the stage microphones located on the balcony, a built-in low-pass filter of 150 Hz were selected to decrease the ventilation hum from nearby vents.

System tuning was commenced by equalizing the sound pressure levels from each loudspeaker individually at the receiving point at the center of the stage. After the calibration, proper configuration for the stage system were applied by acoustical measurement iterations. The same procedure was repeated for the system in the auditorium part of the hall. During the tuning, several combinations of output gain, reverberation time and delay parameters were tried out until satisfactory results were reached. An important step was to add a delay of approximately 90 ms for the hall system and 60 ms for the stage system for creating more natural reverberation. Final parameters for the systems are listed in Table 5.2. In addition, reverberation time at high frequencies were enhanced by applying slight

Table 5.2: Reverberation system parameters in Louhi hall.

	RT_{60}	output gain (linear)	Delay
Stage system	1.5 s	4.7	60 ms
Hall system	2.5 s	3.0	90 ms

equalization above 4 kHz.

5.4 Expectations

The electro-acoustic system used in the small rooms was expected to perform moderately well, depending on the efficiency of the additional room absorption. The walls in the room Cage was not entirely covered with the absorptive curtains, which was anticipated to possibly present some problems.

Louhi hall, with more comprehensive installation and adjusting procedure, was tuned to rather good-sounding hall. The impression of an orchestra was, however, difficult to predict because the subjective opinions in the setup phase were based only on single sound sources. The audible effect of the electro-acoustic system in the audience area was negligible, but this was not considered as a major problem since the system was intended only for the musicians on the stage area.

Chapter 6

Results

In this chapter, the objective results from the acoustical measurements in the case study rooms are presented. To study the acceptance of the system in small rooms, subjective responses from music students and teachers were collected with a questionnaire. The questions concentrated on the perceived acoustical space, loudness of the sound and the enjoyment of playing with the system. Also some opinions about the practice rooms in general were inquired.

A group of nine brass instrument students of varying experience and their teacher participated in the user evaluation for Cage. Disappointingly, only single responses from a clarinet student and teacher were received concerning the room Ives. Three people were interviewed afterward to obtain more comprehensive impressions about the system.

In the Louhi hall case study, a symphony orchestra Tapiola Sinfonietta agreed to rehearse for one day in the virtual acoustic environment. After the rehearsal, a questionnaire similar to the small room cases was asked to be filled by the musicians. To obtain slightly more perspective to the answers concerning Louhi hall, similar questions were asked after their next rehearsal in Tapiola hall about their native rehearsal and concert hall. The questions for Tapiola Sinfonietta were directed toward the ease of ensemble playing with the system and to have comparing analysis between virtual acoustics and good, traditional acoustic conditions.

6.1 Measurement arrangements

The acoustical measurements in small practice rooms were performed within the PureData environment [56] in an automatic playback-recording sequence integrated to the user interface. The only additional equipment needed in the measurements was an omni-directional microphone for signal recording. This microphone was identical to the system microphones. The excitation signal was a logarithmic sine sweep in a single channel.

Room responses were recorded from three source and receiving locations with each acoustic conditions resulting in 54 sweeps in one room. Conversion from deterministic sweeps to impulse responses was performed by using a theory by Farina [18] and the obtained impulse responses were analyzed with the help of IRMA routines for Matlab developed by Peltonen [50]. All the following figures present a backward Schroeder integrated decay curve.

Suitable measurement positions in both small rooms were selected by choosing a combination of possible player locations and a set of system loudspeaker positions. Following the measurement basics, the receiving positions should be located in reverberant and diffuse sound field, instead of the vicinity of the loudspeakers. Especially in the smaller case study room Ives these requirements were followed loosely due to the limited room dimensions. In the reverberation measurements of both rooms, effects of a non-diffuse sound field can be seen at low frequencies. At an early stage the rooms were noticed to have very different acoustics.

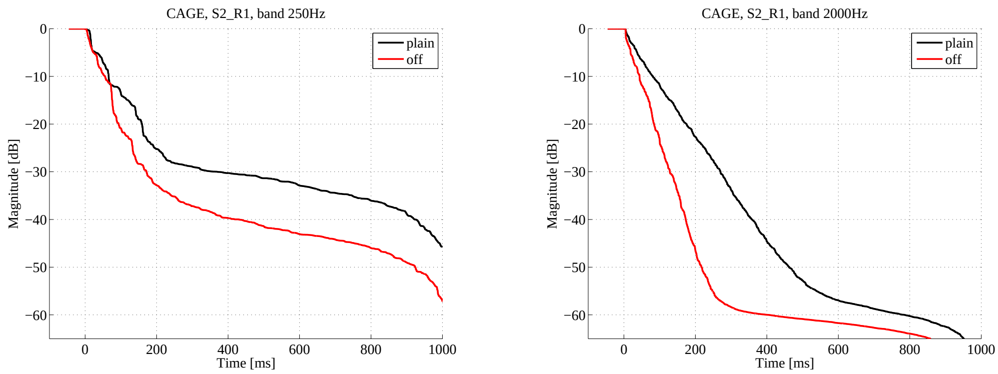
The measurements in Louhi hall were performed with IRMA system [50], and standard measurement equipment consisting of a Bruel & Kjaer 4006 omni-directional microphone and an omni-directional source were applied. Unlike in Cage and Ives, different user selectable settings were not used. Hence, the representation and analysis of the results is slightly different from the small room case studies.

The questionnaire answers are interpreted in the same manner in all case studies. The scale of accordance in each question is evaluated between -10 and 10.

6.2 Case study: Cage

Labeling custom used in figures and tables for different system settings used in Cage and Ives are as follows. An original room without absorption or the acoustic system is referred as *plain* whereas the room with additional absorption installed is called setting *off*. The room with each one of the four different acoustic settings in use are called *on1*, *on2*, *on3*, and *on4* depending of the selected setting. The additional absorption was installed in all these settings. Descriptions for the acoustic system settings were presented earlier in Table 5.1.

Results obtained from the Cage revealed that in its natural state with limited amount of absorption materials, there are certain challenges concerning the sound reflections. The sound decay of the original room (setting *plain*) and when equipped with curtains (setting *off*) at low frequencies, as seen in Figures 6.1(a) and 6.2, is not as smooth as in Figure 6.1(b) at 2000 Hz octave band. This difference results from strong concentration of reflections, which indicates that the sound field was not diffuse at these frequencies in the receiving position in question.



(a) 250 Hz center frequency (b) 2000 Hz center frequency

Figure 6.1: Cage, sound decay, without absorption (curve *plain*) and with absorption (curve *off*), octave bands 250 Hz and 2000 Hz.

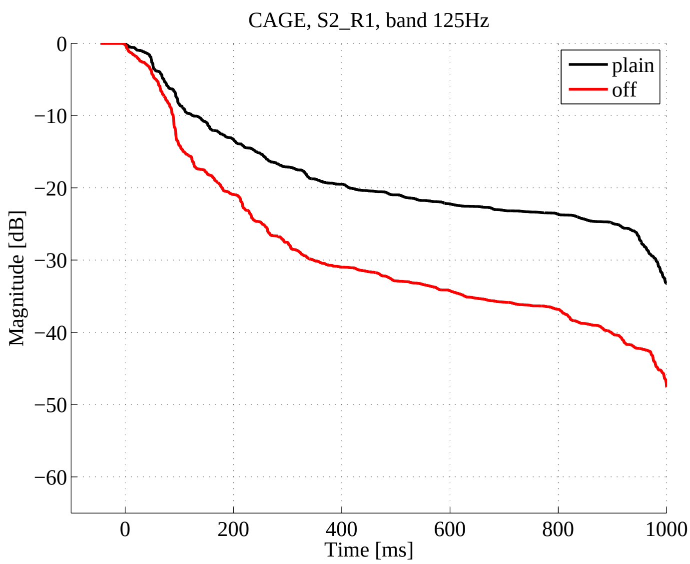


Figure 6.2: Cage, sound decay, without (*plain*) and with absorption (*off*), octave band 125 Hz.

In Figure 6.2 the effect of the absorption on the background noise level is visible, as curve *off* is considerably lower than curve *plain*. Also, both curves indicate that the sound decay is not particularly smooth, which indicates that the sound field is not diffuse at the

125 Hz octave band. The change in the decay rate is, however, noticeable. Similar features is seen in Figure 6.1(a).

At low frequencies, the probability of standing waves between parallel surfaces is rather high. Due to the higher sound energy, these waves can have longer decay times than average. This phenomenon was partially improved by installing the curtains. The possible frequencies of standing waves were predictable from the rectangularity of the room and it's dimensions. Equation (2.15) and the width $L_y = 5.2\ m$ give a result

$$f_{040} = \frac{c}{2}\sqrt{\left(\frac{4}{5.2}\right)^2} \approx 132 Hz \tag{6.1}$$

for the y-axial mode of fourth order. This is near to the 125 Hz octave band center frequency. In addition, the height of the room is 2.75 m, which correspondingly gives $f_{002} \approx 124\ Hz$ by using again equation (2.15). These two axial modes point out a rather strong concentration of characteristic frequencies that fit into the same octave band. Oblique and tangential modes can further emphasize the effect but being less strong their effect is not therefore substantial. Thus, the shape of the decay curves staying at -20 dB and -30 dB at 125 and 250 Hz, respectively, with the setting *plain* is partially explainable by standing waves that remain ringing for unusually long. This effect is less obvious with the absorptive curtains. However, the curtains does not significantly affect to the possible vertical standing waves.

Sound pressure levels

In Table 6.1 relative sound pressure levels on octave bands and A-weighted average are presented. The values are averaged from three source and receiving positions. The room without added absorption is marked as *plain*, damped room as *off*, and damped room with the system in use in the settings *on1* to *on4* with labels *on1* thru *on4* respectively, following the labeling custom in the figures.

At low and middle frequencies, an average reduction of 7 dB in sound pressure levels were obtained by installing the required absorbing curtains, as seen in Table 6.1. Relative changes in sound pressure levels between the different system settings were minor compared to the major drop from the original situation. With the setting *on4*, which was the most audible one, levels were slightly higher throughout the low and middle octave bands while other numbers were within the measurement error. The values for A-weighted sound pressure levels are very similar, mostly being inside the range of approximated measurement accuracy. However, the difference to the original state is undisputed. Relative changes in sound pressure levels were calculated by using Equation (2.35).

According to the user evaluation responses, the drop in general sound pressure level was not clearly perceived. Only two out of ten participants responded that the class under test was quieter than in ordinary state. Four people thought that the room was significantly louder than normally. An interesting dependency can be noticed by combining this result with the other questions: Both respondents with opinion of quieter overall loudness were students with several years of experience. On the opposite side were the teacher, one advanced student but also two players with only little playing experience. During the subsequent interview with the teacher he pointed out that fresh students might have difficulties with their playing technique without the room support they have accustomed to. This can make them try to play harder, thus leaving an overall impression of louder room.

Having the overall A-weighted levels lowered by over 7 dB, a result of participants answering for a clearly noticeable attenuation was expected. However, numbers of this magnitude were not in line with all the subjective opinions.

Table 6.1: Cage relative sound pressure levels in dB, octave bands and A-weighted average.

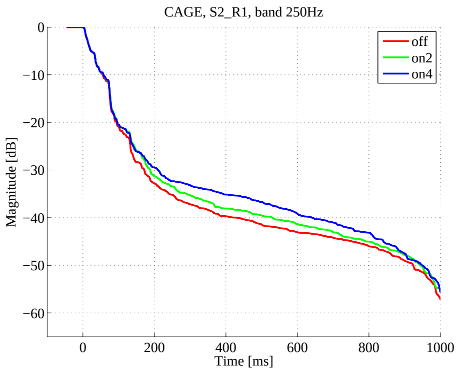
	125	250	500	1000	2000	4000	8000	$L_{w,A}$[dB]
Plain	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Off	-5.9	-7.7	-4.9	-5.6	-9.3	-8.0	-5.0	-7.6
On1	-5.7	-7.0	-4.8	-4.5	-7.0	-7.1	-4.8	-6.8
On2	-5.8	-7.2	-4.8	-5.4	-8.7	-7.6	-4.8	-7.2
On3	-5.8	-6.8	-4.7	-5.1	-8.6	-7.7	-4.9	-7.0
On4	-5.5	-6.4	-4.2	-4.1	-8.0	-7.7	-5.0	-6.5

System responses

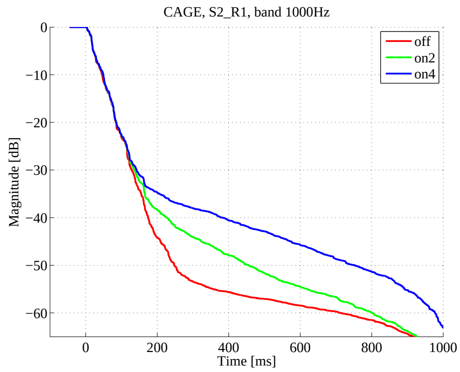
In the class Cage, four presets programmed to the user interface for easy selection resulted in different sound decay behaviors. In the system user interface, settings *on1* and *on2* were labeled *small hall, quiet* and *small hall, loud*, both featuring the same reverberation time of 1.5 s preset in the system and having the only difference in output gain. Settings *on3* and *on4* were similarly labeled large halls with preset $\text{RT}_{60} = 2.4$ s and again having different output gain. At 250 Hz, decay curves for the settings *off* and *on1* are almost identical (appendix B), and the higher output gain of the setting *on2* creates more distinguishable differences compared to the system turned off, shown in Figure 6.3(a). The most audible setting *on4* with the highest output gain and longest reverberation time is also fairly close to the other two settings.

A notable feature with all the settings is that the sound decay is very similar during the first 100 ms. To create more difference in the initial decay, much higher output gains would be needed. Due to the nature of Schroeder integration method, the curves fall sharply when approaching the background noise level. On the other hand, at 1000 Hz band (Figure

6.3(b)) the distinct features between the three selected settings are much more noticeable. Still, the point where the curves start to deviate at approximately -30 dB is far lower than expected by subjective listening. The change of reverberation time in *on2* and *on4* is clearly visible from the slopes after 200 ms. The overall system output gain parameter of these two settings being now set to the same value, it would be theoretically possible to have more similar decay curves at the current octave band by increasing the output gain of the setting *on2* by 7-10 dB.



(a) 250 Hz center frequency



(b) 1000 Hz center frequency

Figure 6.3: Cage, sound decay, settings *off*, *on2* and *on4*, octave bands 250 Hz and 1000 Hz.

The difference between the quiet (*on3*) and loud (*on4*) large hall settings can be seen in Figure 6.4. Decay curves of the settings *on3* and *on4* share the similar shape, while curve *on2* has a different slope due to the shorter reverberation time parameter. This suggests that the parameter change has the desired effect. At 250 Hz octave band, the difference is much smaller.

In Cage, the most used setting among the participants was the quiet large hall (*on3*), while others were in use more rarely. Students who thought their playing sounded better with the system and had generally positive attitude were clearly biased towards the settings with longer reverberation. This suggests that the general attitude towards the experiment would allow to accept unnaturally long reverberation regarding the visual impression and the physical size of the room.

The deviation between the different settings was partly designed to span a wide range so that a favorable parameters for practice acoustics could be found. The quietest setting *on1* had very little support which is understandable - the difference to the system turned off was not clearly audible, even more so when the player should listen to the reverberation during playing. On the other hand, one of the most positive feedbacks from Cage found that this

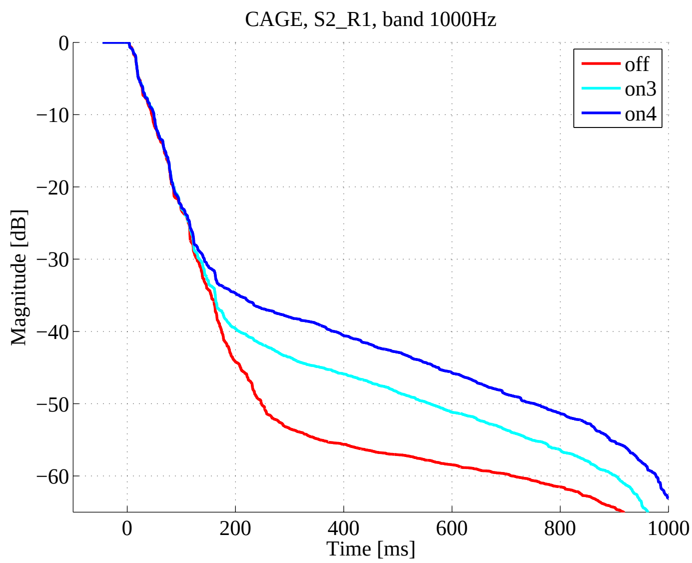


Figure 6.4: Cage, sound decay, settings *off*, *on3* and *on4*, octave band 1000 Hz.

particular setting *on1* was the most natural sounding, as well as the most used setting in this case. The specific participant in question thought also that playing with the system was much more pleasant than without it.

The difference between the two most modest settings is best noticeable at the 1000 Hz octave band (Figure 6.5), although at best there is only 4 dB more gain compared to the system turned off. At lower frequencies the deviation is negligible (Appendix B).

Besides the brass instruments, also a guitar student accompanied by his teacher performed a short testing session in Cage. No written comments were left but feedback about the experience was discussed. Adjustments to the system had to be performed in order to have suitable acoustic response because an acoustical guitar produces much less sound pressure compared to trumpet, for which the system was originally tuned. The output gain parameter was set higher than in the standard settings and reverberation time was set at approximately 3.0 s for easily perceivable results. These participants were interested in the system, but the impression of larger space was seen quite different from real halls. Especially the reverberation at low frequencies was noted to be too strong. The playing position was also closer to the microphones, which could have improved the spatial impression.

Measurements for the system response at these settings were not performed as this was an unexpected situation. However, it is conceivable that the sound decay rates would follow the curve for the setting *on4* in Figure 6.4 with even more modest slope at slightly higher level. The A-weighted sound level with these settings are predicted to be within 1 dB

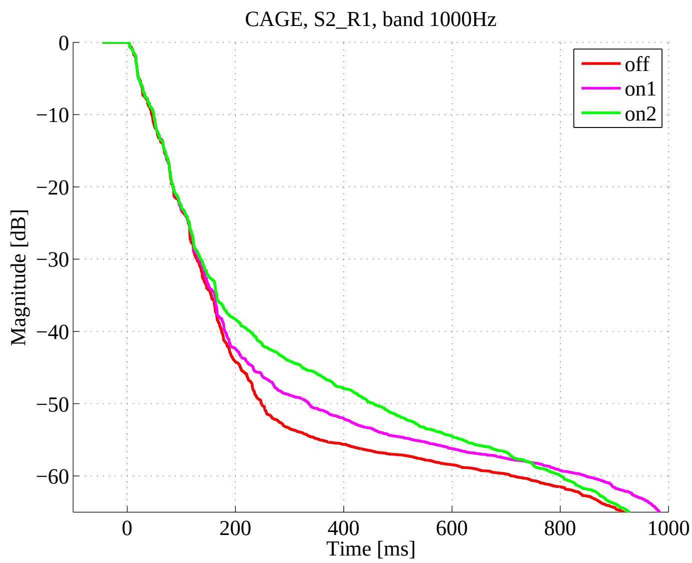


Figure 6.5: Cage, sound decay, settings *off*, *on1* and *on2*, octave band 1000 Hz.

compared to the numbers in Table 6.1. Despite of even a greater increase in total sound pressure levels in the room with the louder settings, an acoustical guitar is not capable to produce uncomfortable sound pressure to the room, not at least by conventional playing techniques.

When trying to define exact values for the measured reverberation times, especially with the system, it is important to notice that the evaluation methods based on graphical analysis pose certain problems. As seen in a number of figures, the curves have a specific dual-decay feature, which means having two separate slopes in different parts of the curve. For instance, calculating RT_{60} for the setting *off* by fitting a line between the points at -5 and -35 dB gives a plausible result, whereas the same method gives false values for the settings *on3* and *on4*. In this case, the reverberation times obtained from the same calculation method would be very similar, neglecting the added reverberation which would still be clearly audible. The straight fitting method is not directly applicable even if the reference points were arbitrary, simply because the curve does not have a slope describing it comprehensively.

User evaluation comments

The testing period in Cage evoked both negative and positive feelings among the respondents. With a question whether one could imagine using a room equipped with a similar virtual acoustic system in the future, the first half of the opinions were rather pessimistic while the others had a more neutral outlook.

One of the trumpet teacher's more precise comments stated the following:

> "Cage is a fairly optimal room for teaching. There is a good amount of reverberation and an adequate sound level. The room should have been damped even more so that different settings could be compared."

In the subsequent discussion this was further pointed out. As Figure 5.1 shows, the whole room was not equipped with absorptive curtains, but instead plain surfaces were left for real reflections, in this case behind the player. Especially for the younger students with yet developing playing technique, the additional early energy could be even more important as supporting factor. For advanced players and teachers having long exposure times, the drop in the general sound levels could, however, be a desired feature.

In Figure 6.6, the shape of the sound decay with the settings *on2* and *on4* can be compared with the empty room at the 500 Hz octave band, which is assumed to be the most comprehensive band for the fundamental frequencies of a trumpet playing range [57]. As the setting *on3* is fairly close to the average of *on2* and *on4*, the teacher's opinion of a good sound level is well based. Still, the loudest setting follows the room's natural decay curve well after 300 ms, but the initial decay is far more rapid with the additional absorption. Despite of the feature, many of the subjects did not appreciate this similarity, and only two of the advanced students mentioned that the setting *on4* was the best for them.

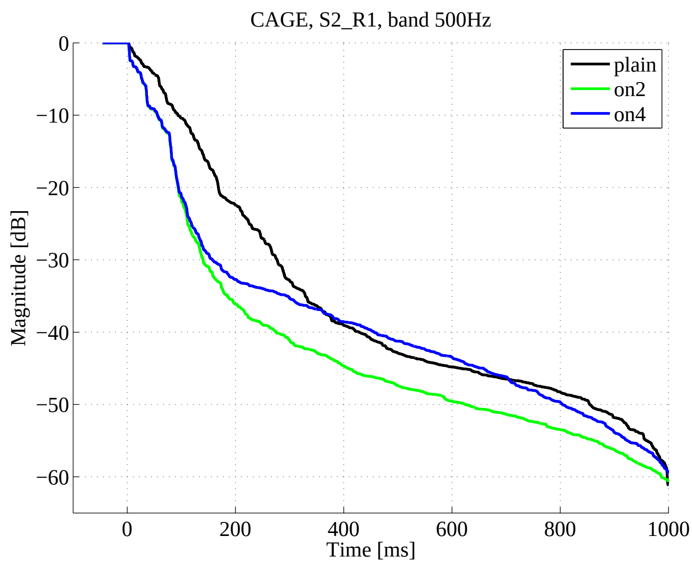


Figure 6.6: Cage, sound decay, original room (*plain*) and settings *on2* and *on4*, octave band 500 Hz.

6.3 Case study: Ives

This section concentrates on presenting the measurement results obtained from the practice room Ives. Subjective feedback is also discussed. Further discussion and distinct comments are found in the next chapter.

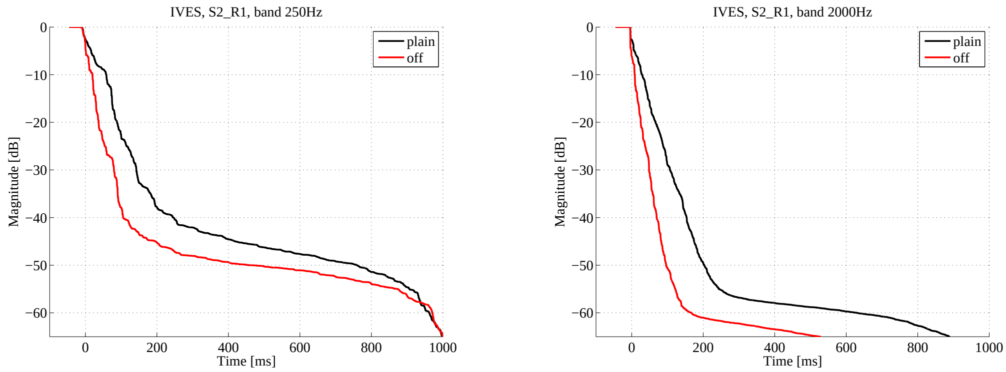
The room provided a short reverberation time of 0.3 s on average without additional absorption. As described in Section 5.2, absorption materials were installed to the room to attenuate the reverberation. Damping a small practice room is beneficial for effectively reducing the reverberation time, although the difference between original room (*plain*) and room with additional absorption (*off*) is not as dramatical as in the room Cage, because the surface materials in Ives are already more absorbing in the original state of the room compared to the materials in Cage. The total effect resulting from the added absorption is the most visible at the 250 Hz octave band, as seen in Figure 6.7(a). If the room had not featured any absorption in its original state, the difference after installing heavy curtains could have been more dramatical, particularly at the high frequency bands. Since there were porous 50mm absorption panels installed, the effect of additional damping is reduced.

The decay curve of the empty room has some fluctuations at the 250 Hz octave band. Similar phenomenon was noticed at the same frequency band in Cage. In this case however, additional absorption smooths the largest irregularities during the first 100 ms. Most likely this change results from rather complete damping of the walls, so that standing waves or other reflection phenomena are attenuated. The possibility of strong reflections between the parallel surfaces is still rather high. The irregular decay curve suggests that the sound field especially without the absorption is not entirely diffuse. The reverberation radius for the empty room is approximately 0.6 m, thus the distance of 1.5 m from the source used in the figures should provide diffuse sound field to the receiving position. Schroeder frequency of the room is 173 Hz.

An interesting observation about the dimensions is that they follow particularly well recommended ratios of room dimensions for optimal distribution of characteristic frequencies. The recommendation of ratios by Sepmeyer [66] is 1:1.14:1.39 while the ratios of Ives are approximately 1:1.14:1.36. The recommended ratio is often used when designing small listening rooms or home theaters.

Another issue for low frequency characteristics relates to the grand piano in the room. A soundboard, located in the bottom of the instrument, works as a resonator amplifying the strings' vibration. It is possible that certain frequencies of the measurement sweep excites the soundboard. According to the studies by Wogram [71], the soundboard modes span the frequencies from 63 Hz to 325 Hz in the case of a concert grand piano, which is considerably longer than the instrument in Ives. Scaling the instrument length of 290 cm from the study to the length of 211 cm in this case raises the mode frequencies slightly, but still covering the 250 Hz octave band. This is a plausible explanation for the difference in the smoothness in Figures 6.7(a) and 6.7(b), since some frequencies on the lower octave band can remain ringing in pianos.

At the 2000 Hz octave band, there is not any signs of irregularities in the decay (Figure 6.7(b)), which shows that the sound field is nearly diffuse, despite of the inevitable proximity of source and receiving positions.



(a) 250 Hz center frequency

(b) 2000 Hz center frequency

Figure 6.7: Ives, sound decay, without absorption (*plain*) and with additional absorption (*off*), octave bands 250 Hz and 2000 Hz.

The difference between the 250 Hz and 125 Hz (Figure 6.8) octave bands is noticeably smaller compared to the same frequencies in Cage. Similar uneven sound decay is still present at the lowest measured octave band in the undamped case, due to the measurement frequency below the Schroeder frequency. The increased wall absorption improves the decay rate as well as the curve smoothness slightly.

Sound pressure levels

Initial impressions for Ives were intimidated by the much smaller size compared to Cage. As a negative side, the limited volume was expected to result in high sound pressure levels. On the other hand, a smaller room was an easier task to dampen more completely. In the questionnaire answers received from Ives, the loudness of the rooms was mentioned to be disturbing.

The sound pressure level drop of similar magnitude was expected in Ives as in Cage by the additional absorption. Both A-weighted and octave band calculations were performed also in this case. In Table 6.2, the effect of the absorption materials and using the electro-

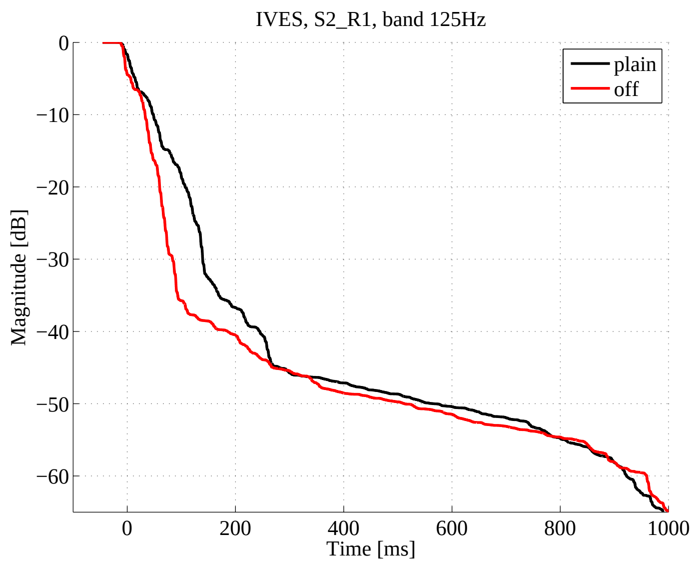


Figure 6.8: Ives, sound decay, without (*plain*) and with absorption (*off*), octave band 125 Hz.

acoustic system is presented by comparing the sound pressure levels in octave bands to the original room.

The most constant change in the sound pressure levels is noticeable at low and middle frequencies, where the levels are higher as the reverberation time and output gain are increased. At 125 Hz, the difference between loudest reverberation and system turned off is 1 dB. One reason for greater increase in SPL at low frequencies is the relative inefficiency of the additional absorption, thus the generated reverberation is more regenerative than at higher frequency bands. The system does not introduce any particular changes in the sound pressure levels above 2000 Hz.

Table 6.2: Ives relative sound pressure levels in dB, octave bands and A-weighted average.

	125	250	500	1000	2000	4000	8000	$L'_{w,A}[dB]$
Plain	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Off	-1.1	-3.2	-2.8	-6.9	-7.5	-7.7	-6.1	-6.8
On1	-0.7	-3.1	-2.7	-6.8	-7.5	-7.7	-6.0	-6.7
On2	-0.8	-3.1	-2.6	-6.8	-7.6	-7.6	-6.1	-6.7
On3	-0.6	-3.0	-2.4	-6.4	-7.5	-7.8	-6.2	-6.6
On4	-0.1	-2.8	-1.9	-5.2	-7.2	-7.8	-6.1	-6.1

System responses

Due to the more compact dimensions of the class Ives compared to those of Cage, some questions arose concerning the proper functioning of the system. After the installation of the absorptive curtains the grand piano was taking up approximately half of the remaining floor area. This would leave the soloist only very little free space, which is a problem especially with larger instruments, such as a cello.

Figure 6.9 shows the overall effect of the system in Ives at the 500 Hz octave band. The early energy is reduced by the additional absorption, but the generated reverberation extends the decay curve beginning from 120 ms. After this the decay is smooth, and with the setting *on2* the level of decay is at best 10 dB higher than in the original room. Naturally the setting *on4* provides longer decay beginning from 60 ms. Total system delay can also be roughly derived from the figure, as the first differences between the settings are present at approximately 30 ms, thus suggesting the initial delay.

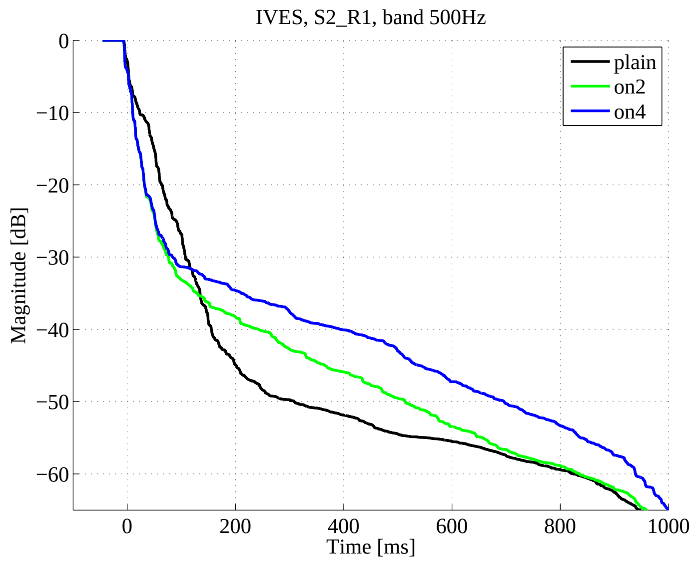
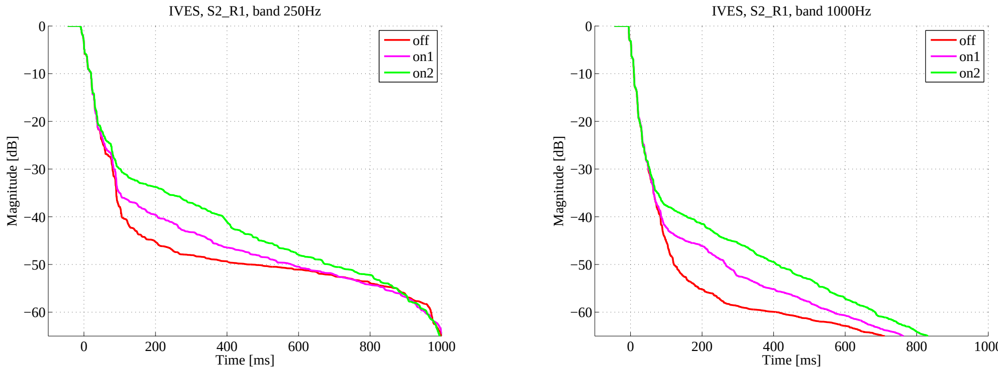


Figure 6.9: Ives, sound decay, original room (*plain*) and settings *on2* and *on4*, octave band 500 Hz.

Figures 6.10(a) and 6.10(b) represent the sound field behavior at the 250 and 1000 Hz octave bands. At 1000 Hz the decay is smooth especially with the setting *on2*. At 250 Hz the decay curves present some irregularities, suggesting that the sound field is not quite diffuse enough. The setting *on1* provided less difference in the decay at 250 Hz than at 1000 Hz. As the curve *on1* follows fairly accurately the average of *on2* and *off*, the output gain setting had the desired effect, which for the setting *on1* was one half of the value for the



(a) 250 Hz center frequency

(b) 1000 Hz center frequency

Figure 6.10: Ives, sound decay, settings *off*, *on1* and *on2*, octave bands 250 Hz and 1000 Hz.

setting *on2*. Such a reduction corresponds to 3 dB decrease in the output sound pressure.

User evaluation comments

Shortly after the testing period in Ives, the participants were interviewed briefly to have more insight to the impressions left by the electro-acoustically enhanced room. The overall opinion of the test was positive. Possible reason for this was the loudness caused by the small size in standard practice rooms. The opinion of the teacher was quite clear about this, and the problem is pronounced with the more advanced players being capable of playing far louder than the less experienced students. The lack of spaciousness and reverberation mentioned during the discussions is generally experienced disturbing. However, this comment refers to the most essential issue addressed in this work - increasing the reverberation but sustaining or decreasing the sound pressure levels. This suggests that active acoustic systems could present a solution for corresponding problems.

The most pleasing setting for the subjects was the louder small hall (*on2*). The quieter option was considered hardly audible whereas the louder large hall was regarded too apparent for this space. The contradiction between visual and auditory impressions in this case was pointed out. A positive result is that the same issue was not mentioned regarding the most favorable setting (small hall, loud), which still provided noticeable improvement in subjective feeling of spaciousness.

> ”The setting for large hall sounded too cathedral-like, it was not suitable for the room in question. The quieter option for small hall were not clearly perceivable.”

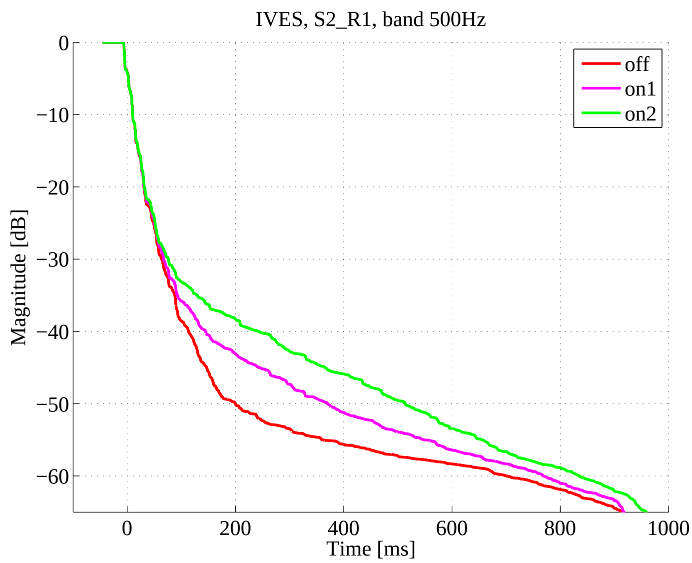


Figure 6.11: Ives, sound decay, without the system (*off*) and on with settings *on2* and *on3*, octave band 500 Hz.

An interesting observation is that according to the teacher, for a listener the setting *on3* of quiet, large hall was more suitable, whereas for the player, the aforementioned setting *on2* was more pleasant.

Differences between the measurement results with different settings were subtle, but adjacent settings were distinguished by the subjects. In Figure 6.11, the decay curves are presented for the settings *on1* and *on2*, of which the first one was regarded rather inaudible while the setting *on2* was found optimal. The difference is not very large, about 5 dB from 200 to 600 ms. In sound pressure levels the difference is negligible.

All in all, the acoustical setting pleasing the subjects most was the one which provided a perceivable impression of larger space that was in proportion with the characteristic dimensions of the room.

6.4 Case study: Louhi hall

While standard measurements for the stage parameters were not practically possible in the small practice rooms, the Louhi hall case study provided valuable results from the effect of the installed system. Reverberation times were measured only on the stage, since the system did not have considerable effect in the audience part of the hall. Due to this limitation, the results are not fully comparable to the measurements from other halls.

Table 6.3: Reverberation times in Louhi hall with and without the electro-acoustic system, octave bands.

	125	250	500	1000	2000	4000	8000	Hz
T_{20}, system off	1.0	1.0	1.0	1.0	0.9	0.9	0.7	s
T_{20}, system on	2.5	2.4	2.3	2.2	1.7	1.3	0.9	s
System effect	1.5	1.4	1.3	1.2	0.7	0.4	0.2	s
Concertgebouw (NL)	2.2	-	2.1	-	1.8	-	-	s
Symphony Hall (Boston, USA)	2.2	-	1.8	-	1.7	-	-	s

An example of the effect of the electro-acoustic system to the time energy curves is displayed in Figure 6.12. The top left figure shows that the reverberation generated by the system has the characteristics of a natural late reverberation. The top right figure presents the first 300 ms of the time energy curve, which is very different with the system on and off. The lower figures show the corresponding decay behavior with only the stage system active and without the high frequency equalization described in Section 5.3. These measurements were performed at the source and receiving positions S1 and P1 that are close to each other. This explains the relatively high energy of the direct sound.

In Table 6.3, the reverberation times T_{20} are presented. The values are averaged from 3 source and 3 receiver positions. At low and middle frequencies, noticeable increase in the reverberation times was gained. For reference, some values from renowned concert halls are shown [57].

By comparing Tables 6.3 and 6.4, the effect of the system can be noticed. The effect on EDTP parameter is much smaller than on T_{20} for the reason that the system is not designed to produce early reflections. For reference, values for good acoustic concert halls vary between 1.5 and 2.0 s (average from 250-1000Hz) [23].

Table 6.4: Early decay times on stage in Louhi hall with and without the electro-acoustic system, octave bands.

	125	250	500	1000	2000	4000	8000	Hz
EDTP, system off	0.9	0.9	0.9	1.0	1.0	1.0	0.7	s
EDTP, system on	1.4	1.7	1.5	1.5	1.3	1.2	0.8	s
System effect	0.5	0.8	0.6	0.5	0.4	0.2	0.1	s

Whether the hall should have less or more reverberation, the musicians found that the reverberation with the system was fairly well suitable for an orchestra practice hall. The average of all answers was 0.4. Compared to a good acoustical hall, the stage and the audience area should give an impression of slightly larger space, as the answers were 1.0

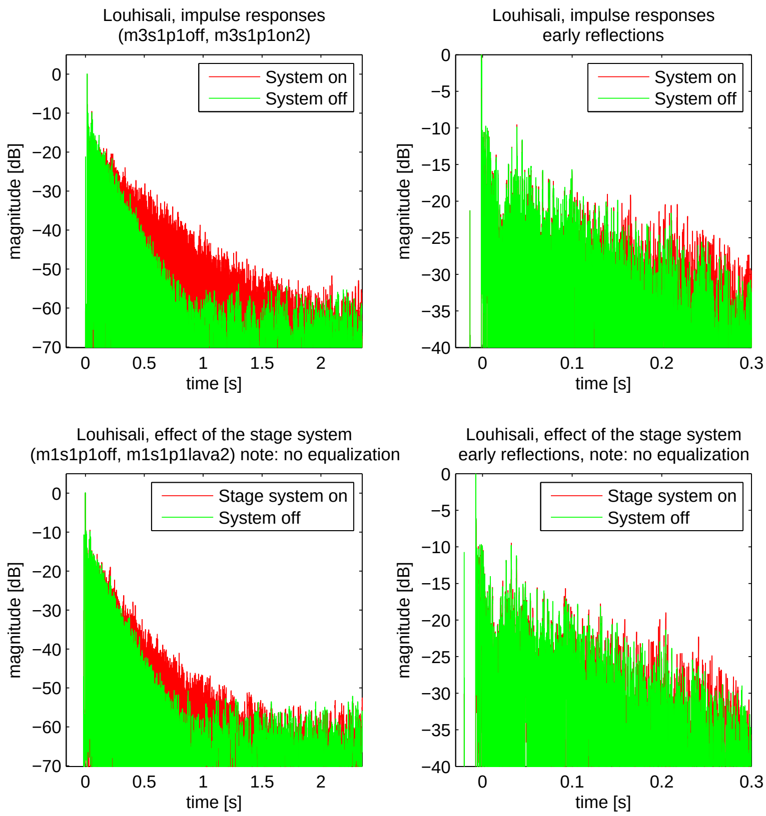


Figure 6.12: Louhi hall, time energy curves with and without the system. In bottom figures only the stage system is in use.

and 1.5 on average, respectively. However, there was a bit too much late reverberation arriving from the audience according to the musicians, the average value being -0.5. Still, all these results are fairly well neutral, which suggests that there were no dramatical flaws in the system. Also the standard deviations of the answers were at most 2.5.

On the other hand, more support for ensemble playing was desired. All instrument categories were quite unanimous about this question since there were not any opinions complaining about too much support, while the other bassoonist described that ”there can never be too much support”. The average for this question was 2.7. However, the overall ensemble playing in Louhi hall with the system was very highly regarded, the averaged rating being as high as 5.5.

Support parameter ST1 is used to objectively describe the ease of ensemble playing. In Louhi hall, the average value from the 250-1000 Hz octave bands was -8.3 dB (Table 6.5),

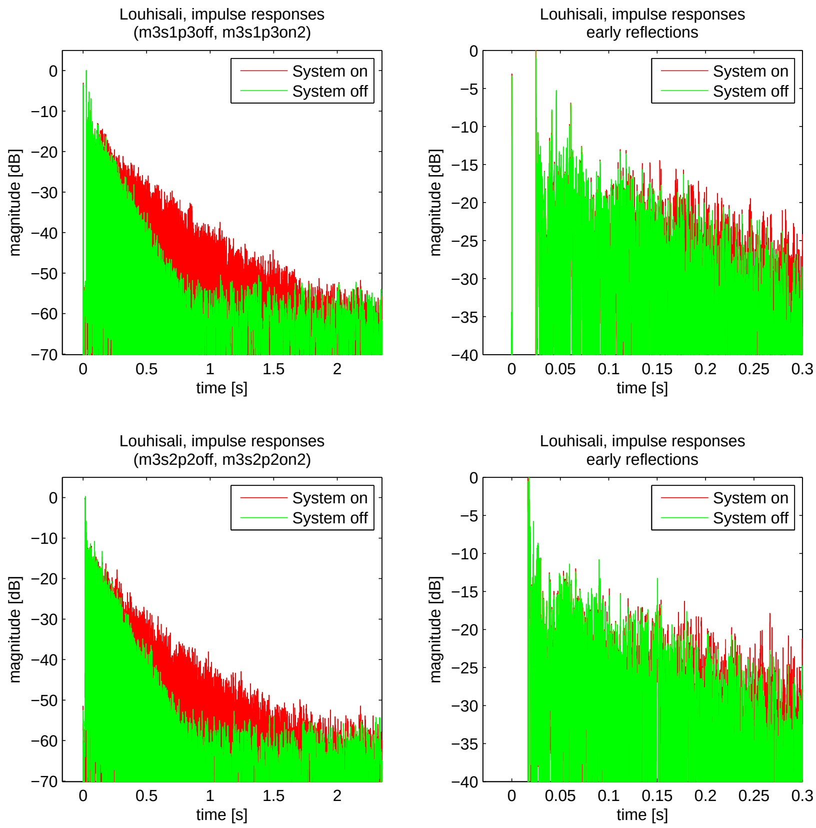


Figure 6.13: Louhi hall, time energy curves with and without the system.

while according to Gade [23], the highest corresponding value in European halls is -10.9 dB. The difference is noticeable and supports the answers from the musicians. Gade has mentioned an ST1 value as high as -9.7 dB in a theater orchestra pit, but this was also considered too loud. On the other hand, this problem was not present in this study. The overall rise in the sound pressure levels with the system switched on was clearly below 1 dB at all octave bands between 125 and 8000 Hz (see Table 6.6). As for the support parameter ST2, slightly higher results were rather obvious, since the early energy is not increased with the system.

Gade has proposed that the ST2 values should be 1-3 dB higher than the values for the ST1 parameter. Again, the average difference at the 250-1000 Hz octave bands is 1.8 dB, which is very much on the suggested range.

The clarity values on the stage were lower with the system than without electro-acoustics. This is connected to the addition of late reverberation. The measured values are shown in Table 6.7. In the investigations by Gade [23], from the octave bands 250-1000 Hz the optimum average values of 12-13 dB are cited, while in known halls CS values of 13.7 (Musikverein) and 14.2 (Concertgebouw) are recorded. The corresponding average in Louhi hall with the system was 13.6. However, values as high as 17-18 dB were found in other European halls by Gade. Questions directly related to the impression of clarity was not asked in the questionnaire, but it is plausible to suppose that the clarity values are closely related to the unexpectedly good results for opinions concerning ensemble playing. Figure 6.14 shows the Louhi hall before and during the rehearsal. For reference, the questions and answers are presented in Appendix A.

Figure 6.14: Louhi hall, on the left with the system ready to use, Tapiola Sinfonietta in rehearsal on the right.

Table 6.5: ST1 (early) and ST2 (total) support values in Louhi hall with and without the electro-acoustic system, octave bands.

	125	250	500	1000	2000	4000	8000	Hz
ST1, system off	-5.7	-9.9	-6.7	-8.4	-6.7	-7.7	-9.1	s
ST1, system on	-5.8	-9.9	-6.8	-8.3	-6.7	-7.7	-9.2	s
System effect	-0.1	0.0	-0.1	0.1	-0.1	0.0	0.0	s
ST2, system off	-4.6	-8.0	-5.6	-6.9	-5.5	-6.4	-8.3	dB
ST2, system on	-4.5	-7.6	-5.4	-6.5	-5.4	-6.3	-8.2	s
System effect	0.0	0.4	0.3	0.4	0.1	0.1	0.0	s

Table 6.6: Sound pressure levels in Louhi hall, octave bands. SPL figures are relative to the hall without the system.

	125	250	500	1000	2000	4000	8000	Hz
SPL, System effect	0.4	0.7	0.6	0.5	0.3	0.2	0.1	dB

Table 6.7: Clarity on stage values in Louhi hall with and without the electro-acoustic system, octave bands.

	125	250	500	1000	2000	4000	8000	Hz
CS, system off	13.2	15.1	15.5	15.4	15.2	15.0	20.0	s
CS, system on	11.8	13.5	13.4	13.8	14.2	14.3	19.5	s
System effect	-1.3	-1.6	-2.1	-1.7	-1.0	-0.7	-0.4	s

Chapter 7

Discussion

The measurements performed in case study rooms provided valuable results of the system behavior. In the next sections, the subjective opinions as well as possible causes and improvements for some issues are discussed. For the Louhi hall case study, the responses gathered from the musicians are compared to the questionnaire results obtained in Tapiola hall (Appendix A).

7.1 Case study: Cage

The properties of the instrument - more closely the sound pressure level, dynamics and directivity - were noticed to be surprisingly important for making a small practice rooms suitable for music rehearsal. The brass instruments, of which the trumpet and the cornet were used in Cage, represent a category with comparably narrow dynamic range but high sound level and directivity [20, 49, 57].

Directivity properties for the used instruments can be found problematic with the system installed in a room like Cage. With a setup using two microphones placed along adjacent walls, the distance between these receiving positions was large. The gap between the microphones was covered by curtains that absorb the sound energy. Normally the sound would reflect back from the plain wall. Thus, when the instrument is pointed between the microphones, part of the sound is attenuated before the reflections reach the player again. The floor and ceiling being now highly reflective compared to the area covered with the absorbing material, the direction of the reflections and feedback is changed. For some users, this might result in a situation where the system itself is generating ambiguous ambiance instead of real reverberation. However, by installing absorptive and diffusing elements also on the floor and at the ceiling, the system reverberation could be more easily distinguished. The overall output gain of the system could be then increased, but only to a certain limit due to the coloration and acoustic feedback problems.

In the discussions after the testing period in Cage, one reason considered for the effect described above was the low height of the room and insufficient floor and wall absorption. According to the teacher, the undamped areas did produce reverberation of their own in some degree so that the abilities of the installed system could not be fully used. The comment suggests that the surround-like approach is probably not the best solution for equipment positioning with instruments having high directivity. This indicates that for more directive instruments, the loudspeakers and microphones should be positioned to the playing direction instead of around the musician. As a consequence, the artificial reverberation would appear to arrive from the same direction and provide more support.

Some complaints were also addressed to the partially audible localization of some speakers. The main concern was the loudspeaker situated utmost to the left from the playing position. This speaker was also the closest to the teacher while a student was playing. Therefore the output level of the loudspeaker in question had to be slightly lowered after testing the system for a while. Half of the answerers claimed that there were not any perceptible localization, also the very same respondents expressing that their playing did sound better or similar with the system but not worse. In addition, the same set of answers stated that the system reverberation did not contradict with the natural reverberation of the room. The issue of localization was improved for the room Ives with custom tuned delay lines for each output channel to equalize the arrival times from each speaker to the principal playing spot while maintaining equal output levels.

From the visual perspective, the thick, black curtains installed in any ordinary room make the surroundings look smaller and less spacious. This can be seen as a contradiction causing an unnatural reaction towards reverberation properties dissimilar to the visual appearance [5]. Perrott [51] has stated that visual stimuli is regarded more important than those of auditory system's. This phenomenon lowers the possibility of credibly using settings that create an impression of noticeably larger spaces. This rather psychological issue is not directly belonging to the scope of this work and is not analyzed further.

By paying more regard to the instrument directivity and the echoic properties of the acoustics in the room where the subjects were accustomed to practice, more positive general opinions could have been received in Cage. As the optimal solution in this case would offer plenty of early energy to provide the responsiveness for the ease of playing, the room should have been equipped more completely with absorptive materials and correspondingly increase the output gain of the system. However, this would easily result as disturbing coloration in the sound.

An alternative for this method would be to distribute the absorption materials evenly throughout the room and leaving small sections without absorption. By adjusting the relation of plain walls and the absorption, a satisfactory amount of early energy could be attained while slightly reducing the sound pressure levels. After all, this is close to the principle of truncated hall concept, but only in a small room.

7.2 Case study: Ives

Altough a very small number of participants took part to the Ives case study, some good impressions were gathered in the interviews. Similar concerns of loudness and support for inexperienced students were discussed as in Cage. Advanced players, such as the participant in Ives, do not have problems practicing in acoustically dry rooms. On the contrary, fresh students might find such an environment uninspiring and cause unnecessary stress due to the apparent lack of usual acoustical support, even if the enhancement of late reverberation would give the impression of a larger hall.

Issues of lacking responsiveness - similar to those encountered in Cage - were not experienced in Ives. As the clarinet is more omnidirectional than the trumpet, the clarinet sound is distributed more evenly to the room. Regarding to this theory, the clarinet tends to create more easily a diffuse sound field, which could provide more room support than in the case of the trumpet. This would also make the clarinet a more suitable instrument for small rooms. The slightly different playing position of the clarinet is as well a possible cause for the difference in perceived support. Conventional trumpet playing position causes the sound to radiate to more forward direction, while in Cage a lot of absorption was positioned in this direction. Hence, the sound energy normally reflected from the playing direction was reduced.

It is also possible that the subjects tended to directly associate the impression of support to higher sound pressure levels. However, the decrease in sound pressure levels with the additional absorption did not receive complaints as the subjects found that the practice rooms were generally too loud.

In the subsequent discussions, the reduced sound pressure levels were regarded an extremely positive feature, especially according to the clarinet teacher. Although in the questionnaire answers this was not clearly perceived, the change was mentioned afterward. Multiple practice rooms were described as intolerably loud during long exposures. As the dimensions are similar in most practice rooms, it is possible that some room modes are excited by the clarinet sound. Hence, an increase in the sound pressure and sound coloration can be perceived in certain playing positions. This issue was addressed in the interviews and supports the assumption above.

The drop of over 6 dB in A-weighted levels compared to the original room corresponds roughly to a decrease of the noise dosage to one-fourth from the original, which is a notable improvement for hearing protection. Supposing that an average daily noise exposure for a music teacher is 3 hours in 80 dB, also the temporary hearing threshold shift is noticeably lower with the decrease of 6 dB in sound pressure level. Correspondingly, the recovery from threshold shift is more rapid in this case. As the differences in the sound pressure levels measured in Cage were similar, comparable deductions are valid for also Cage. [28, 57]

The problem of audio-visual contradiction existed in Ives as well as in Cage. According to the subjects, this effect was more apparent with longer reverberation settings as the visual and aural impressions were further apart. During the testing, other less obtrusive alternatives for efficient absorption were not sensible. For longer periods, installation of absorption material with less striking appearance would be plausible. Hiding the loudspeakers would be a more difficult task, as the sound quality should be high enough, thus restricting the minimum loudspeaker size and positioning.

7.3 Case study: Louhi hall

The subjective opinions gathered from the Louhi hall case study provided interesting results with the comparison to the answers received from Tapiola hall. The biggest difference between these two halls was the quality and ease of ensemble playing. A second matter is related to the opinions from both halls when compared to a good or ideal acoustical hall.

In section 6.4, the values for stage parameters were presented. As the support parameters ST1 and ST2 (Table 6.5) were very high compared to other halls and to the suggested support parameter values [23], the opinions evaluating the ease of ensemble playing were extremely positive. In contrast, the corresponding situation in Tapiola hall was regarded much worse compared to the electro-acoustically enhanced Louhi hall. The difference between the opinion averages was 9.0. In this matter there were also slightly less inconsistency between the individual answers in Louhi hall than in Tapiola hall, standard deviations being 3.4 for Louhi hall and 3.8 for Tapiola hall. Regarding the conditions on the stage of Tapiola hall, several written comments were received as follows:

> ”Listening to the other instruments is harder than in Louhi hall.”

> ”Certain instrument groups are difficult to hear.”

Unfortunately, the ST1 and ST2 support parameters were not available from Tapiola hall for establishing a direct comparison. One answer also pointed out that objective evaluation is impossible due to the ”deafness” to the acoustics in a very familiar hall. Instead in Louhi hall, the written responses for the ensemble playing were enthusiastic apart from few reservations:

> ”Intonation improved a lot by this system!”
>
> ”The system clearly improved the mutual audibility.”
>
> ”All instrument groups could be heard, although own playing could not be distinguished entirely sufficiently.”

In the questionnaires presented in both halls, the musicians were asked to compare the specific hall to an acoustically good hall (in Louhi hall) and to an ideal hall (in Tapiola hall). Despite of the verbal difference, the same goal was sought with these questions. Some accusations related to the needs for acoustical improvement in Tapiola hall have already been under discussion.

To sound even more like a good traditional concert hall, the most required feature in Louhi hall was more stage support with the average of 2.7. By the measurement values, it is highly possible that the musicians have never played in a spacious sounding room with that much support on the stage. ST2 parameter average at 250-1000 Hz was measured at -5.3 dB while an ideal value suggests approximately -10 dB values. Out of various European concert halls, -10.1 dB is the highest measured ST2 value [23]. Despite of this fact, the musicians wanted still more support.

Other matters, such as loudness, reverberation and the impression of size were regarded fairly optimal. A little too much late reverberation from the audience was perceived as the average was -0.5 for sufficient reverberation. A single answer complained about the excess reverberation of the grand piano, which is explainable by the absence of the lid. This caused the sound of the piano to radiate rather directly to the system microphones (see Figure 6.14.

The lack of stage support and the response from the audience area was considered the greatest deficiency in the stage acoustics of Tapiola hall. In all received questionnaires, more support was required with an average value of 6.1. The response from the audience was regarded insufficient with an average of 3.9. A frequent suggestion among the answers was an improvement by installing reflecting clouds or canopy above the stage to provide additional reflections from the ceiling.

In all the questions for comparing Tapiola hall to a hall with ideal acoustics, the standard deviation of the answers was surprisingly high, and a difference between instrument types were noticeable. Especially the violin and the viola sections required more impression of a larger stage whereas wind instrument sections wanted a slightly smaller space. Because the Louhi hall case study was arranged to be held on the first rehearsal day of the autumn season, the lack of very recent experiences of traditional halls might have had a slight effect on the results as well. The high standard deviation suggests that the constant playing positions on the stage have developed varying preferences for different stage acoustics, possibly due to the difference between the properties of the instruments.

The standard deviations for the corresponding questions in Louhi hall were much smaller, typically only a half of the values in Tapiola hall. This indicates that the musicians have formed their own outlook after playing in the hall for many years, which is not the case with Louhi hall and the electro-acoustic system. Therefore a more comprehensive testing period would be necessary to obtain more determined opinions. Nevertheless, some references can be derived by comparing the answers from different instrument sections. String players were generally the most pleased with the performance of the system. On the contrary, brass and woodwind players located more on the back of the stage were not fully satisfied due to the apparent loudspeaker localization. Most of the discontent was caused by their relatively distant position from all the system microphones and correspondingly the proximity to the hindmost loudspeakers on the stage balconies (Figure 6.14). This caused the system to have less supporting effect on these positions while allowing the musicians to hear other instruments more clearly.

The time-variance in the algorithm was not perceivable in the negative sense during the orchestra rehearsal. However, a slight fluctuation was heard while adjusting the system previously. Also a problem was encountered as the system fluctuation at selected modulation frequency interfered with the piano tuning process.

In Figure 7.1, the sound decay and the early reflections can be compared between two measurement position pairs. The top figures show the response form source S1 to receiver P3, while the bottom figures present the response from source S2 to the same receiver (see Figure 5.5(a)). The measurement position pair S2P3 corresponds with the acoustical impression in the back row of the stage. Figure 7.1 does not present any marked differences between the positions. However, the sound source used in the measurements was omni-directional, while brass instruments tend to have stronger directivity features. This affects to the response from the system.

The sound pressure levels presented in Table 6.6 are not expected to pose any actual risk regarding the potential hearing loss even on the long-term compared to the same hall without the system. Issues related to a healthy sound environment are most likely caused by the physical structures in the practice space instead of the active acoustic system. On the other hand, the reverberation in a hall of the size of Louhi hall without the system could not be much shorter without losing the natural reflections, which the system at this point is not able to simulate.

7.4 Future improvements

The electro-acoustic system utilized in the case studies of this work provided a good, flexible environment for enhancing the reverberation in practice and rehearsal rooms. Of the

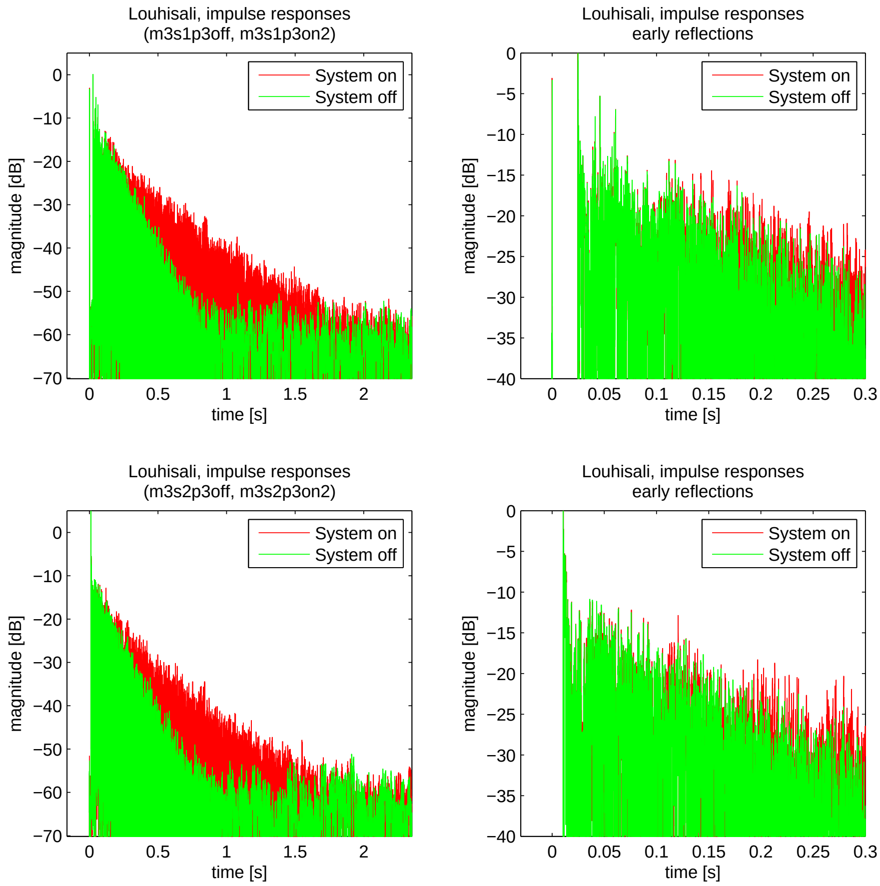


Figure 7.1: Louhi hall, time energy curves with and without the system.

technical implementation and features, a more customizable low-pass filter in the reverberation algorithm (depicted in Figure 4.1) would improve the capabilities in fine-tuning the system. A related issue was encountered in the Louhi hall case study while adjusting the timbre of the reverberation. This problem was solved by searching suitable settings for a parametric equalization, which was implemented to the signal processing algorithm. Therefore a low-pass filter with adjustable slope would be preferred.

An even more sophisticated alternative would be a modified reverberation algorithm with the characteristics of a graphical equalizer. In this case the reverberation time and output gain could be adjusted individually for each octave or thirds band. Such adjustments have been earlier featured for instance in ACS system [7]. By using a traditional band-pass filtering for the implementation, the signal processing load increases in proportion to the number of bands.

In Louhi hall with two separate systems, the tested environment had only an acoustical coupling between the systems. With additional channels it would be possible to directly connect the systems to provide more realistic reflections normally occurring in concert halls. While both systems had their different, subjectively tuned reverberation parameters, feeding some of the hall system output to the stage system inputs could generate more late support for the musicians. This would have only a small impact on the system stability. With the experiences on the system resiliency against coloration, this should not pose any difficulties.

A whole different approach could be designed with wave field synthesis [30]. With several loudspeaker arrays, the system could reduce the loudpeaker localization especially in small rooms. Downsides for this kind of systems are considerably more complicated implementation and installation in addition to more extensive needs for equipment, thus also larger spaces. In small practice rooms, such as Cage and Ives, this would not be practical.

The setup of two separate systems could be used also in smaller spaces than Louhi hall. A minimum floor area of the room for such an installation would be approximately 120 m^2. Firstly, the rooms of this size would allow to position the equipment at sufficient distance from the musicians. Secondly, the natural reverberation of such a room is in most cases adequate to permit the use of additional delays in the artificial reverberation. In Louhi hall, these delays were seen to markedly improve the quality of the generated reverberation. The installation would feature different systems for simulating the stage and the audience area. With a careful system tuning, this concept could provide an environment for smaller ensembles with less than 10 musicians.

An important area for research in the future is an efficient solution for the sound pick-up with different instruments. As described earlier, microphone positioning has a significant effect on the quality of the electro-acoustic system. With the current microphone knowledge, adjustment of the system is required depending on the instrument. Therefore an optimal solution would remove this problem.

Chapter 8

Conclusions and Future Work

In this work the use of electro-acoustic systems for enhancing the acoustics in music practice and rehearsal spaces has been studied. Two typical small practice rooms were investigated in addition to the conversion of a small multipurpose hall to a rehearsal space for a symphony orchestra. In each case, subjective opinions were gathered with a questionnaire to study whether virtual acoustic systems could be used as a part of acoustical design in practice spaces.

The main idea behind the investigations was to increase the total absorption in a closed space to decrease reverberation times but also the sound pressure levels, and to add the reverberation with an electro-acoustic system. As a result, the room would appear to be acoustically larger than in reality.

The system utilized a time-variant reverberation algorithm in order to increase system stability and to produce as natural reverberation as possible. The small rooms were treated with an absorbing drapery sustained on lighting supports and bars. The room acoustics in the symphony orchestra rehearsal hall investigation was already sufficiently dry without additional absorption.

The results obtained in the small rooms showed that the sound pressure levels were considerably lower with the system installed than in their original states. In both rooms the A-weighted sound pressure levels were about 6 dB lower with the system independent of the reverberation settings. Not even the use of excess reverberation did have any notable effect on the levels.

Of the four available settings, the reverberation parameters featuring RT_{60} of approximately 1.5 s and clearly distinguishable output gain was regarded the most useful and natural sounding option. Half of the subjects in small rooms considered that similar systems could be used in practice rooms in the future. Psychological aspects were also found to have an effect on the acoustical impression.

In the symphony orchestra rehearsal hall case, the system was tuned so that the subjects did not have any control to the system. With the system, the reverberation time increased from 1.0 to 2.3 s at middle frequencies while other stage acoustic parameters were brought to similar values with renowned European concert halls. On the other hand, the sound pressure level increase introduced by the system was below 0.7 dB at all octave bands.

The system was received very positively by the musicians. Only four out of 30 musicians responded that such a system could not be used in orchestra rehearsals. The hall equipped with virtual acoustics was regarded a good rehearsal space and the improvement over the same hall without the system was remarkable.

As these investigations have proved, practice rooms can be modified by an electro-acoustic system. Improvements in small rooms require different installations depending on the instruments and the acoustical properties of the original room. Hence, additional research in the fields of small practice rooms and microphone positioning is needed. In larger spaces the problems are mainly related to providing sufficient effect to all positions on the stage area. Also the importance of careful fine-tuning of the system was found essential.

The future research on this subject should concentrate on modifying the used reverberation algorithm to provide more customizable decay in frequency domain. An interesting advance would be the possibility to interpret previously recorded impulse response from an existing acoustical space and automatically adjust the reverberation parameters to comply with the principal acoustical features. In the studies concerning virtual acoustics in small practice rooms, a sufficient number of experienced subjects should be ensured over a longer time period.

> ”The operation of such a system requires personnel having a sense of musical appreciation with regard to levels and balance of the various elements.”
> -J.W. Ditamore, 1965 [17]

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Appendix A

User evaluation questionnaires

Questions of virtual acoustics in Louhi hall

1. Compared to Louhi hall with the acoustic system, a rehearsal hall for the orchestra should sound: - quieter or louder? (-10 - +10)
2. - drier or more reverberative? (-10 - +10)
3. - smaller or larger? (-10 - +10)
4. *Comments*
5. To make the acoustics of Louhi hall more like a good hall, Louhi hall - the stage should sound quieter or louder? (-10 - +10)
6. - the hall should sound quieter or louder? (-10 - +10)
7. - should support ensemble playing less or more? (-10 - +10)
8. - should give an impression of a smaller or larger stage? (-10 - +10)
9. - should give an impression of a smaller of larger hall? (-10 - +10)
10. - should create late hall reverberation less or more? (-10 - +10)
11. Do you find that the ensemble playing was particularly hard or easy in Louhi hall equipped with the system? (-10 - +10)
12. *Comments*
13. Were there any discrete directions from where the system sound was arriving, not at all / very clearly? (-10 - +10)

14. *Comments*
15. Did the reverberation from the system interfere with the characteristic reverberation in the hall, not at all / very much? (-10 - +10)
16. *Comments*
17. If you have played in Louhi hall earlier, was playing now with the system more pleasant, less enjoyable / more enjoyable? (-10 - +10)
18. *Comments*
19. Did the sound of the orchestra appear to be better with the system than in a similar space without a system, worse / better? (-10 - +10)
20. *Comments*
21. Could you think the orchestra to have rehearsals in a hall equipped with similar system, not at all / definitely? (-10 - +10)

Questions of acoustics in Tapiola hall

1. Compared to Tapiola hall, a rehearsal hall for the orchestra should sound: - quieter or louder? (-10 - +10)
2. - drier or more reverberative? (-10 - +10)
3. - smaller or larger? (-10 - +10)
4. *Comments*
5. To sound like an ideal acoustical hall, Tapiola hall - the stage should sound quieter or louder? (-10 - +10)
6. - the hall should sound quieter or louder? (-10 - +10)
7. - should support ensemble playing less or more? (-10 - +10)
8. - should give an impression of a smaller or larger stage? (-10 - +10)
9. - should give an impression of a smaller of larger hall? (-10 - +10)
10. - should create late hall reverberation less or more? (-10 - +10)
11. Do you find that the ensemble playing is particularly hard or easy in Tapiola hall? (-10 - +10)
12. *Comments*

Table A.1: Louhi hall questions and answers.

#	instr.	1	2	3	5	6	7	8	9	10	11	13	15	17	19	21
1	1. vl	0	-2	-1	0	0	0	0	-2	-4	9	-10		9	10	5
2	1. vl				10	0	10	9	9	0	10		-10	10	10	10
3	1. vl	-2	-3	-2	-2	-1	3	0	2	1	0			5		5
4	1. vl	1	1	4	0	0	3	1	2	-1	5	0	-7	9	9	8
5	1. vl	3	0	0	2	2	6	0	0	-2	10	-5	-10	10	10	10
6	1. vl	0	0	0	0	0	0	0	0	-2	10	-10	0	10	10	6
7	1. vl	-2	0	1	-1	-1	2	0	1	-2	5	-9	-9	9	6	0
8	1. vl	0	5	5	0	0	0	0	0	-2	5	0	-5	5	5	5
9	2. vl	0	0	0	0	0	2	2	2	2	5	-8	-6	9	6	6
10	2. vl	0	0	1	0		2	1	0	1	6	-10	-9	9	7	5
11	2. vl	1	6	5	0	0	3	4	4	-4	8					
12	2. vl	0	0	0	0		0	0	-1	1		-10	-10	10	5	6
13	2. vl	0	-1	0	0		2	-2	-1	-2	5	-8	-8	7	3	5
14	vla	1	1	1	0	1	1	1	2	1	5		-10	5	3	4
15	vla	3	3	3	3	1	2		3	2	5	-7	-10	8	9	8
16	vla	-4	3	4	3			3	5		0	-10	-10	10	10	10
17	vla	0	0	0					5		10	-10	-10	10	10	10
18	vc	-2	0	0	-1		2	0	0	-1	5	-9	-9	8	8	8
19	vc	-2	-2	-2	0		2	0	1	2	8	-9	-8	8	8	5
20	vc	-2	-2	-1	0	-2	2	-2	-2	-2	5		1	2	2	-4
21	cb	2	-1	0	1	4	9	1	3	2	3	-6	-6	2	3	-2
22	fl	-1	0	0	-2		4	0	0		4	-9	-9			
23	cl	-2	-1	0	0		2	2		-2	-4	-7	1	2	2	-10
24	ob	-2	2	3	0	2	3	3	3	-3	1	-10	-5	10	10	10
25	bsn	0	-2	0	0	-2	2			-4	10	2		10	10	-4
26	bsn	0	1	1	0		5	0	1	2	5	-10	-10	10	10	10
27	tr	0	0	0	0	0	0	0	0	-4	8	-2	-6	10	10	9
28	tr	0	-1	-1	0	-1	3	1	-1	-1	5	8	0	10	10	5
29	corno	3	5	5	2	2	2	2	3	3	5	5	-5	9	9	4
30	timp	-2	-1	0	0	2	4	1	4	6	6					

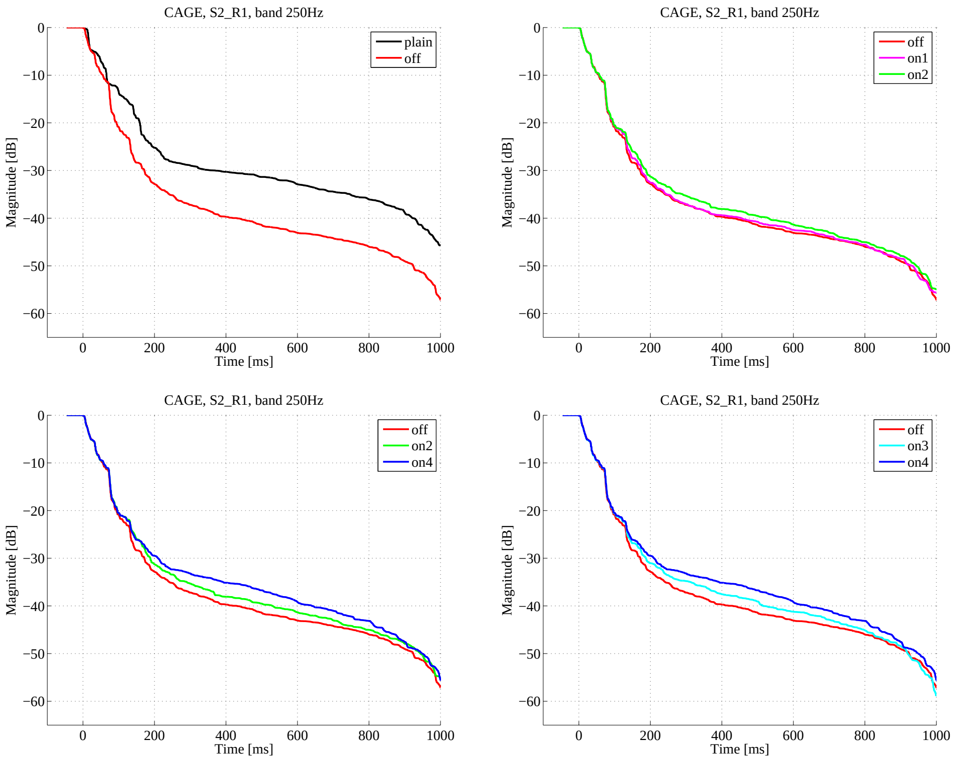
Table A.2: Tapiola hall questions and answers.

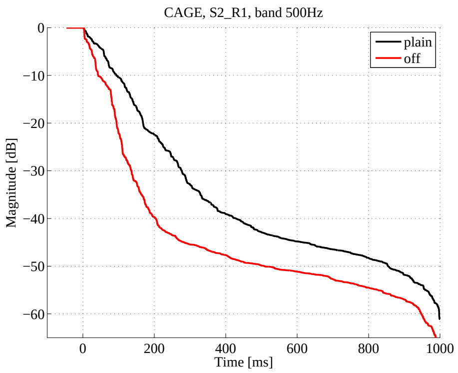
#	instr.	1	2	3	5	6	7	8	9	10	11
1	1. vl	2	2	1	5	10	10	4	8	10	2
2	1. vl	0	0	10		0	10	10	10	10	-5
3	1. vl	0	1	0	0	5	5	0	0	2	-2
4	2. vl	-10	-10	0	2	0	10	10	4	2	-5
5	2. vl	2	4	1	2	9	4	2	3	8	-2
6	2. vl	2	1	1	1	8	7	1	2	6	-5
7	2. vl	-2	-2	-2	-3	3	4	-3	5	-8	-5
8	vla	-2	3	0	-2	10	10	10	10	-10	-10
9	vla	-1	2	0	0	-1	3	1	1	0	2
10	vla	-2	-2	-2	2	10	10	-5	-10	5	-6
11	vla					3	6				-4
12	vc	0	0	0	0	5	4	-3	4	4	0
13	vc	1	0	0	0	3	3	2	0	4	-2
14	cb	0	3	2	2	5	4	-2	-3	2	-3
15	cb	2	-1	-1	2	3	8	-4		3	-1
16	fl	0	0	0	-2	0	2	0	0	0	5
17	ob	-4	0	-1	3		4	-2	0	-2	-3
18	bsn	10	10	10	-3	0	3	-5	0	5	-10
19	bsn	3	3	-5	1	3	10	0	-2	5	-9
20	tr	0	0	0	0	0	5	1	-2	5	-4
21	tr	0	0	0	0	0	10	-3	0	0	-6
22	timp	-2	-3	-1	-2	6	3	-3	0	-2	-4

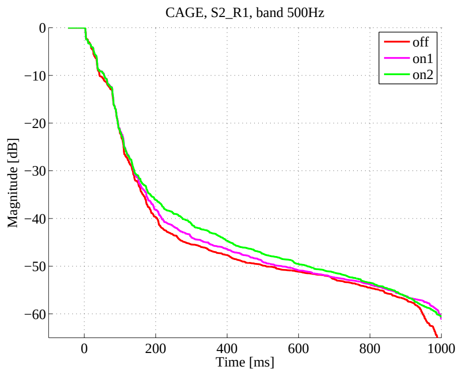
Appendix B

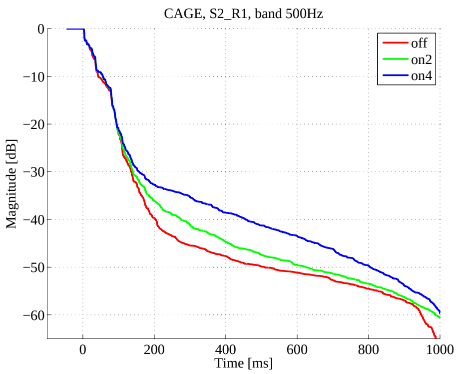
Additional reverberation figures

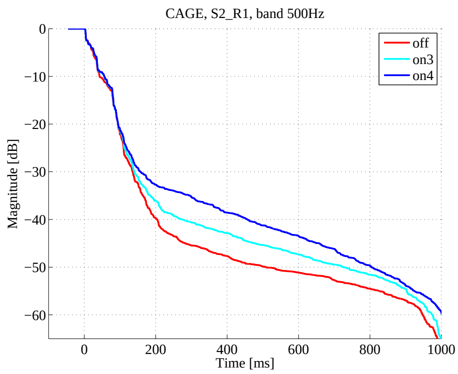
Cage

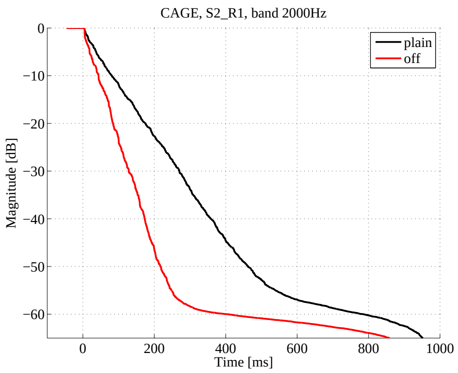


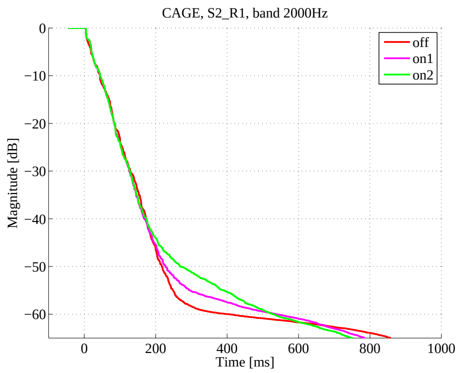

CAGE, S2_R1, band 500Hz
plain
off
Magnitude [dB]
Time [ms]

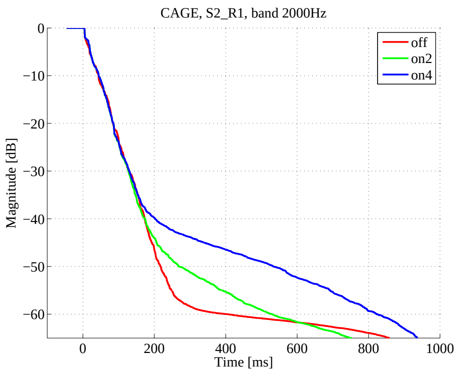

CAGE, S2_R1, band 500Hz
off
on1
on2
Magnitude [dB]
Time [ms]

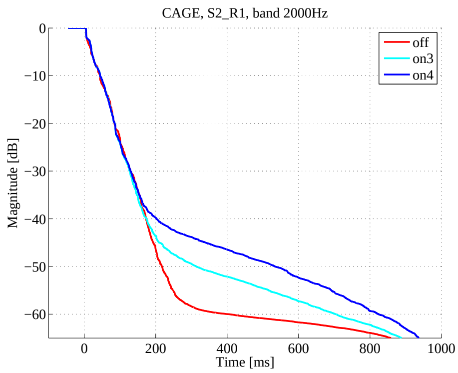

CAGE, S2_R1, band 500Hz
off
on2
on4
Magnitude [dB]
Time [ms]


CAGE, S2_R1, band 500Hz
off
on3
on4
Magnitude [dB]
Time [ms]

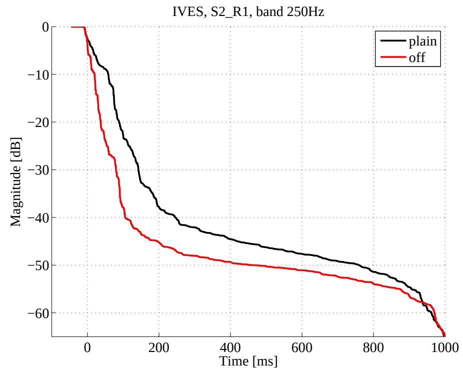

CAGE, S2_R1, band 2000Hz
plain
off
Magnitude [dB]
Time [ms]

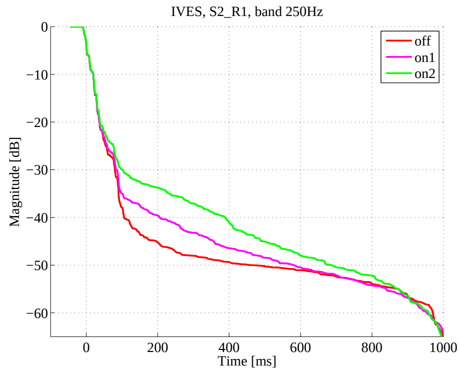

CAGE, S2_R1, band 2000Hz
off
on1
on2
Magnitude [dB]
Time [ms]

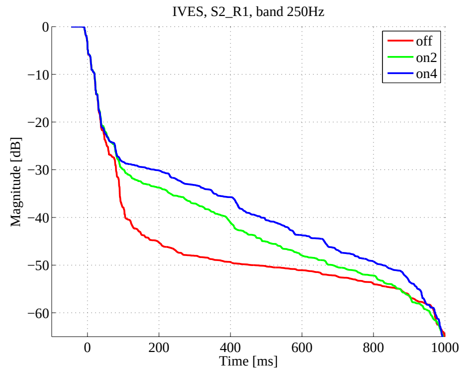

CAGE, S2_R1, band 2000Hz
off
on2
on4
Magnitude [dB]
Time [ms]

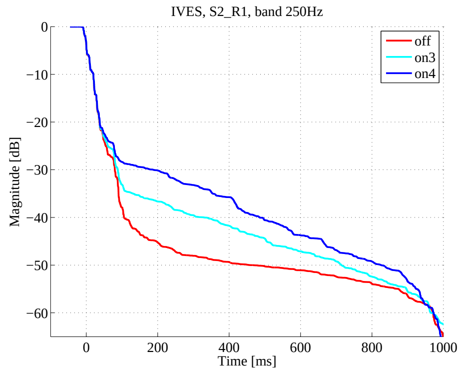

CAGE, S2_R1, band 2000Hz
off
on3
on4
Magnitude [dB]
Time [ms]

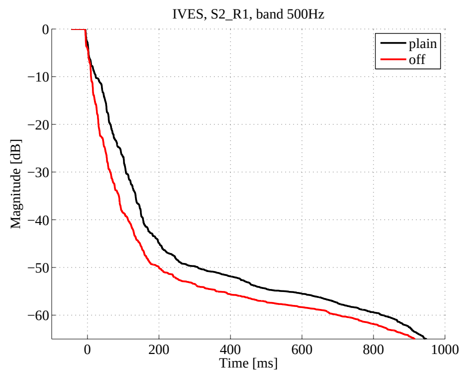
Ives

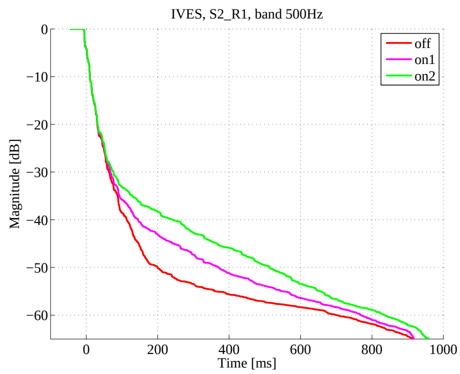

IVES, S2_R1, band 250Hz
plain
off
Magnitude [dB]
Time [ms]

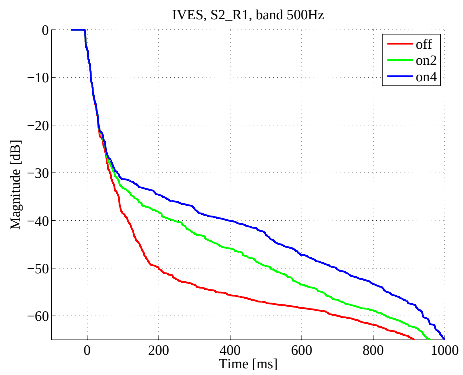

IVES, S2_R1, band 250Hz
off
on1
on2
Magnitude [dB]
Time [ms]

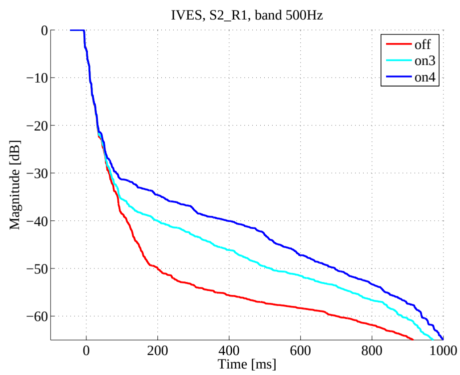

IVES, S2_R1, band 250Hz
off
on2
on4
Magnitude [dB]
Time [ms]

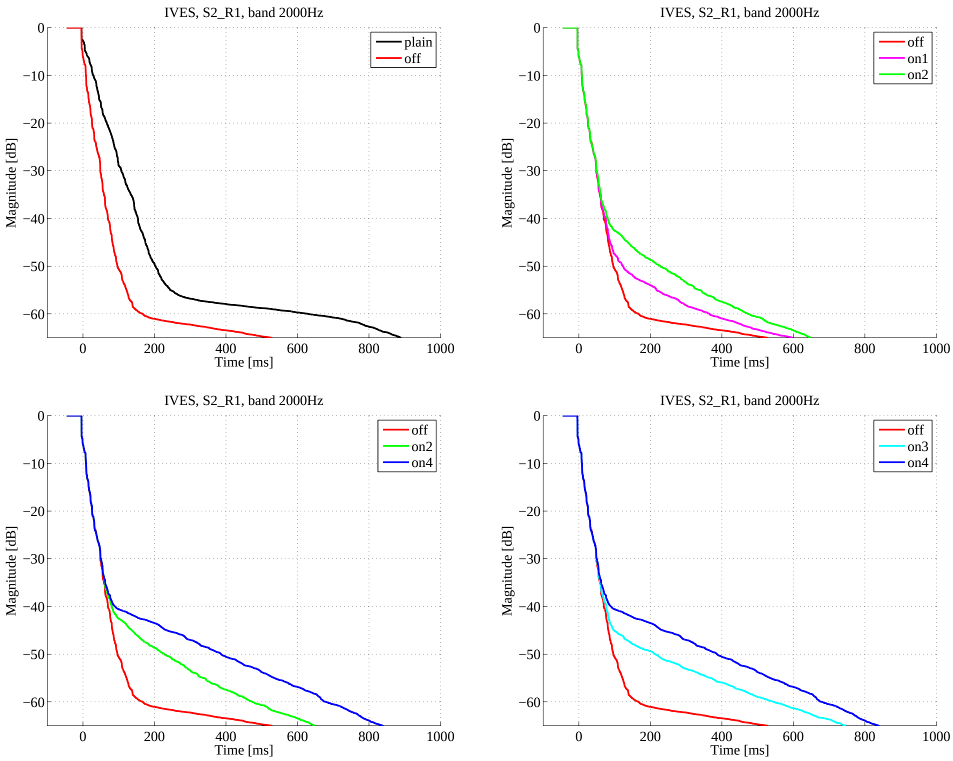

IVES, S2_R1, band 250Hz
off
on3
on4
Magnitude [dB]
Time [ms]


IVES, S2_R1, band 500Hz
plain
off
Magnitude [dB]
Time [ms]


IVES, S2_R1, band 500Hz
off
on1
on2
Magnitude [dB]
Time [ms]


IVES, S2_R1, band 500Hz
off
on2
on4
Magnitude [dB]
Time [ms]


IVES, S2_R1, band 500Hz
off
on3
on4
Magnitude [dB]
Time [ms]


IVES, S2_R1, band 2000Hz
plain
off
Magnitude [dB]
Time [ms]
IVES, S2_R1, band 2000Hz
off
on1
on2
Magnitude [dB]
Time [ms]
IVES, S2_R1, band 2000Hz
off
on2
on4
Magnitude [dB]
Time [ms]
IVES, S2_R1, band 2000Hz
off
on3
on4
Magnitude [dB]
Time [ms]