Acoustics in Practice

International e-Journal of the
European Acoustics Association (EAA)

Vol. 2 • No. 1 • January 2014

Acoustics in Practice®

International e-Journal of the European Acoustics Association (EAA)

Vol. 2 • No. 1 • January 2014

SUMMARY

P. 3
An aid to innovation in instrument-making: a collaborative approach between workshops and research laboratories
Doutaut Vincent, Curtit Marthe

P. 13
Musician´s perceived timbre and strenght in (too) small rooms
Tor Halmrast

P. 31
The effect of scattering objects on measured reverberation times in sport halls
Don Oeters

P. 35
Extracting meaningful uncertainty data from Calibration Certificates and associated Sound Level Meter Standards
Ian Campbell

P. 43
An eye-witness report on how the CD came about
Jacques Heemskerk

editor in chief
Colin English

editorial assistant
Monika Rychtarikova

editing coordinator
Miguel Ausejo

edited by
European Acoustics Association (EAA)
secretary@european-acoustics.net • office@european-acoustics.net
www.euracoustics.org
c/o. Sociedad Española de Acústica (SEA)
Serrano, 144, ES-28006 Madrid, Spain
Legal Deposit: M-21922-2013 • ISSN: 2226-5147
Printing by DiScript Preimpresión, S. L.

www.euracoustics.org

An aid to innovation in instrument-making: a collaborative approach between workshops and research laboratories

Doutaut Vincent, Curtit Marthe
Itemm – Institut technologique européen des métiers de la musique, Le Mans, France
Corresponding author: vincent.doutaut@itemm.fr

PACS: 43.75.-z

ABSTRACT

The economics of French instrument-making is mostly made up of very small handicraft enterprises. But craftsmen do not always have the means to embark, by themselves, on an innovation strategy. The idea is to try to respond to current challenges in instrument making; for example, the reduction in costs of design times or the adaptation to customer needs. All of this necessitates the development of low cost tools for characterising and prototyping of instruments dedicated to their use in workshops. Examples of collaborative approach between instrument makers and research laboratories are presented. Mainly, the PAFI project, for "Instrument-making Aid Platform" aims to develop characterisation tools for all the instrument families. The initiative's originality lies in the fact that "pilot craftsmen" are associated with every stage of development. The process involves a research program and the support of craftsmen for developing the hardware and software. Bearing in mind the international economic context, these experiences may act as a basis for broadening and pursuing this initiative on an international scale. The open and progressive nature of the work means that we can consider such a prospect with a view to maintaining the small-scale production of high-quality instruments.

1. INTRODUCTION

The economics of French instrument making consists almost entirely of very small handicraft enterprises. Faced with strong international competition, they are positioned on the top-end and concert musical instruments market, while learning instruments are rather manufactured on an industrial basis. This observation may certainly be applied more generally at European level, and even at international level.

The craft industry production of musical instruments is the result of a long process of learning know-how and optimising the instrument making stages. The development of traditional instruments is an ongoing process that occurs in response to the needs of musicians and composers (both contemporary and historic ones), or is imposed by constraints on the supply of protected materials, such as certain types of exotic wood. Consequently, the use of alternative materials, the modification of structural elements, and improvements made to intonation or acoustic instrument radiation are now carried out using scientific tools. While this approach is widespread for industrial applications, this is seldom the case within handicraft businesses.

The issue, therefore, is to develop tools for characterising instruments suitable for use in a workshop so that instrument-makers may have the means at their disposal for innovating by keeping their production quality one step ahead and their handicraft enterprises competitive.

How a response can be provided to the geographic and thematic dispersal of the craftsmen? The objective is to present some initiatives taken in France over the last few years in order to bring instrument-makers and research laboratories closer together in a structured, sustainable way and create a joint working dynamic. This illustration is provided based on the PAFI project (Instrument Making Aid Platform) – which is intended, for the medium and long term, to provide *low-cost* tools suitable for workshop operations and concrete responses to situations involving the innovation of musical instruments. What makes the process original is the fact that instrument-makers are involved in all the development stages, defining the specifications and testing the prototypes.

This paper aims to present an original approach to scientific and technical collaboration between a collective of craftsmen, manufacturers and repairers of musical instruments, a grouping of research laboratories, and a technology transfer and support structure for enterprises: Itemm, “Institut technologique européen des métiers de la musique”.

This paper outlines the environment and the special conditions which led to this experiment. The issue of innovation as a part of instrument making is specific to the French context and does seem to have the capacity for general application at international level. So what is involved here is a presentation of examples of technology transfers that contribute to maintaining craft industry production of high-quality musical instruments. The paper shows how to provide instrument-makers with tools for innovating and improving instruments which are suited to their needs, and which are the outcome of scientific research over the last twenty years, particularly in the field of musical acoustics.

2. ECONOMIC BACKGROUND

Instrument making belongs to the cultural industries and now the leisure industries (technically speaking, engineering and broadcasting of musical signals) which, by their very nature, require in-depth analysis of “musical sound” and the mastery of the cutting-edge technologies associated with it. Instrument making, by rights, constitutes one of the renowned economic sectors drawing directly on the results of scientific research, and particularly research in the field of musical acoustics and material sciences.

2.1. The music industry

The advent of electronic and digital technologies is bringing about far-reaching, drastic changes in the music industry and, consequently, instrument making, to the extent of deliberately using electronic accessories and even computers, as fully-fledged musical instruments. Historically speaking, the starting point for this change was the recurring issue of amplifying instruments which, over the last century, have been the outcome of new cultural practices related on the one hand to the growth of Afro-American music and then electro-acoustic music

Figure 1. *Skilled-craft industry sector: services all around repair and maintenance of musical instruments, from manufacture to direct selling.*

and, on the other hand, to the change in places where performance are provided and music is listened to, which applies to all types of music.

A consequence of this technological and cultural revolution has been a change in the landscape in terms of instrument making. From the second half of the 20th century onwards, the mass-manufacturing of musical instruments gradually moved from Europe to the United States of America, Japan, South-East Asia and now, China. This distribution is also to do with an industrial mode for instruments development (electro-acoustic and electronic instruments, among others) which began in the USA and was applied to electric guitars. It now impacts on all the instrument families, where a few major brands dominate the market, particularly in the case of electronic instrument making. Computer-assisted music and sound synthesis technologies provide examples of this perspective.

Along with the expansion of amplified forms of music, music-making is made accessible to all and worldwide cultural intermixing and have also brought about this change in the manufacturing sites for traditional acoustic instruments on the beginner and learner markets. For example, only instrument making that provides high added value has been retained in Europe, with a few "specialities", like the piano industry in Germany and the wind instrument industry in France.

2.2. The skilled-craft industry sector

Although in this case the presentation is restricted to the typology applicable to French instrument making, it seems that the clear distinction between international industrial companies on the one hand and a network of very small handicraft enterprises on the other hand, reflects the situation in many countries.

It is quite difficult to obtain an accurate picture of the French instrument making sector. An overview of its economic activity leads to a current assessment of the number of companies involved as being around 2,500 (the manufacturing, restoration, repair, maintenance, tuning, servicing and marketing of musical instruments), providing a pool of about 10,000 jobs [1]. These figures tell us a great deal, because in order to fully delineate this instrument sector, all of the actors involved in the technical field must be taken into account: industrials, craftsmen, distributors, and music stores of a general or specialised nature with a repair workshop.

Musical instruments production, which is positioned at the top end of the market, now constitutes the output more or less solely of very small handicraft enterprises with fewer than twenty employees, two-thirds of which are sole proprietorships. What is actually entailed in the sector is no longer just craft-industry or semi-industrial manufacturing, but the existence of structures engaged in "mixed" technical and commercial lines of business, offering repair and maintenance services along with customer advice and direct selling. Pools of rental instruments, which consequently require regular technical maintenance, are becoming increasingly important within this context (markets for learning instruments, particularly for young people receiving musical training).

The standard profile for a craft industry workshop, to a varying extent depending on the instrument speciality, is thus oriented towards a service offer ranging from the manufacture and repair of instruments (sometimes to a very high level of technical quality and know-how) through to marketing and maintenance, and even, increasingly, the finishing and adjustment of imported unfinished instruments, within the learning instruments market niche. The estimated number of companies with a repair workshop is between 1,600 and 1,800. The estimated number of enterprises with a recognised manufacturing line of business is nearly 800.

3. TECHNOLOGICAL AND ENVIRONMENTAL CHALLENGES

From a technical perspective, the basic characteristic of instrument making is the multidisciplinary nature of the skills which, quite often, are held by the same person within the company:

- Use of multiple materials (woods, metals, skins, textiles, composites);
- Mastery of related types of know-how (design, assemblies, finishes);
- Varied technical and scientific knowledge (of a mechanical, acoustic, or electronic nature, or relating to computer-integrated manufacturing);
- Various types of cultural knowledge (music, organology, instrument playing);
- And, of course, company management (supplies, inventory, trading, communication).

And this is not to mention the advent of electronic instruments and computer music technology The international economic context is now imposing ever-greater integration of technological innovation, which handicraft enterprises are far from being exempt from. Quite to the contrary, keeping ahead in terms of quality demands (both on the products and the services that they offer) or, in other words, in relation to both "top end" and "customised" items, involves the intense development of appropriate technical solutions and their quick implementation within the enterprises, following the example of the developments noted over the last thirty years in mass-production units in all sectors. The benefits derived from this are also the same:

- *Reduced costs and shorter design and execution timeframes* (virtual prototyping, selection of materials, reproducibility, production monitoring and control, etc.);
- *Capitalising on knowledge and practices* (products traceability, the optimisation and development of models, etc.);
- And, a key advantage for small enterprises; *adaptation to customers' needs* (response to specific demands, responsiveness, service quality, etc.).

Moreover, as in the case in a number of other business sectors, instrument making is also confronted with changes to standards and new regulations concerning workplace health and safety, or which relate to the themes of procurement and sustainable development. By way of an illustration, one of the major issues currently relates to the natural materials normally used for manufacturing musical instruments. Going above and beyond just the framework of restoring instruments that are collectors' items, international agreements for protecting flora and fauna concerning certain endangered species (CITES see [2]) are now adversely affecting the production of new instruments: the pernambuco used in modern bows and the rosewood used in guitars and percussion instruments, for example. In the medium-term, the precondition for maintaining a competitive craft industry production of high-quality musical instruments is not only seeking alternative high-performance materials (wood essences and/or composites), but also determining appropriate dimensions for their use in instrument making.

Thus, *combining the maintenance of traditional know-how and the use of innovative methods and means* currently lies at the heart of the challenges to be taken up by the instrument making sector [3]. In order to provide an effective response, we need to overcome the isolation of craftsmen and provide solutions that are of common interest to each family of instruments.

4. INNOVATION AND INSTRUMENT MAKING

The relations between music and science go back a long way. Instrument-makers have continually sought to understand how the instruments that they make work so that they can improve their performance and adapt their characteristics to what musicians want. Moreover, advances made in research have already enabled significant spin-offs in terms of the characterization and sizing of instruments, and prediction and prototyping tools. Currently, instrument-makers even have products dedicated to instrumentation such as BIAS and VIAS, developed by IWK [4]. The initiatives presented here are not positioned as complementary tools. What is original about them is the collaborative approach adopted in defining them, which involved the instrument makers themselves.

4.1. Skilled-craft industry/research dynamics

Since 2001, annual meetings between luthiers and scientists have been organised by the Itemm with the support[1] of one of the main French professional associations, the UNFI (National Instrument Making Union). These meetings firstly devoted to stringed instruments[2], which are held over a two-day period, provide the opportunity to exchange information via specialised lectures devoted to musical acoustics, materials sciences (particularly "resonance" woods used in instrument making), as well as tutorials applying vibration and acoustics measurement equipment. These simulations provide an educational framework within which scientific ideas and concepts can be discussed, criticised, understood and adapted to the specific features of the instruments.

Among the various examples of tutorials carried out, the following may be cited for example: identification of the initial resonances and categorization of guitars using vibration measurements or determination of the wood characteristics based on the resonances of the bow sticks. From 2001 to 2007, many experiments were designed and presented. Their design and development involved more than sixty students from the acoustics training courses offered by Le Mans University (France). This potential enables responding to the concrete issues raised by craftsmen.

These interactions have been more widely and progressively extended to the whole French scientific musical acoustics community, implying particularly PhD student researches. Each current biannual meeting – one for string instruments and the other for wind instruments – has thus two main objectives: on the one hand a regular information about recent research results and, on the other hand, an agenda

[1] The sessions are also supported by the musical acoustics group of the French acoustical society (GSAM-SFA) since 2008.

[2] On the same basis, there are from now on such annual meetings devoted to wind instruments since 2010.

Figure 2. *Biannual meeting participants (string and wind instruments): instrument makers and scientists joint working dynamic.*

and a better definition of solutions to be found to efficiently help makers. Therefore, this dynamic contributes to the elaboration of a collective solutions and produces a common strategy aiming for global innovation improvement within the workshop.

4.2. Individual support

After six years of collaborative effort (2001-2007), the closer links between luthiers and university staff resulted in the development of a pilot metrology tool called *Lutherie tools*, developed by the "Laboratoire d'Acoustique de l'Université du Maine" (LAUM) and the "Pôle d'innovation des métiers de la musique" (music professions innovation centre[3]), Itemm. This system was made available to craftsmen free of charge, in exchange for feedback about using the tools. Above and beyond being a tool with predefined functionalities enabling a response to the needs identified. The main purpose of *Lutherie tools* was to provide luthiers and bow-makers with guidance concerning their thinking and their approach to characterizing their work.

As an experimental device, *Lutherie tools* consists of various equipment items (vibration and force sensors, signal conditioning box) and analysis software (developed with RAD framework Matlab® and distributed as compiled executables). It is intended for stringed instruments and is organized into specialized modules: the guitar, harp, quartet and bow. By measuring transfer functions, it enables instrument elements, and even whole instruments, to be characterised and sized. The objective here is not to provide details of the system's various functionalities, but rather to present the project's development approach which was chosen [5].

Lutherie tools has an open design in order to respond to the issues specific to each instrument, and to enable developments of the various functionalities to be oriented according to the feedback obtained. One of the major advantages of the system is its low cost when it comes to ensuring broad dissemination within workshops. While some of the system's functionalities are already to be found in commercial products used in the field of mechanical engineering, they are only available piecemeal and their complexity requires a level of expertise that most instrument-makers do not have.

Ten or so copies of the initial prototype of the device were produced in 2007, and then twenty or so in 2009 for the second version. The systems are loaned at professional meetings for a one-year period. Modifications are made based on the critical feedback provided. These loans are extended by being associated with a personal applied research project, formulated by each of the instrument-makers involved in the dynamics. Each individual project is supported and monitored over the current year and forms the subject of a presentation of the results at the next professional meeting. It has to be noticed that this individual support is essential to facilitate the adjustment of the educational process to the real needs of the makers and to obtain more detailed feedbacks to improve iteratively the forthcoming field studies and actions.

5. THE PAFI PROJECT

The PAFI (French acronym for Instrument Making Aid Platform) project, drawn up over a four-year period (2009-2013) and may be seen as the general

[3] The name "Pôle dinnovation" (innovation centre) is awarded subject to a French Ministry for the Craft Industries quality label granted to an establishment for developing an economic and in particular technological support mission for a business sector. The Itemm, one of the main instrument making training centres at European level, is in charge of technology transfers for companies within this field in France.

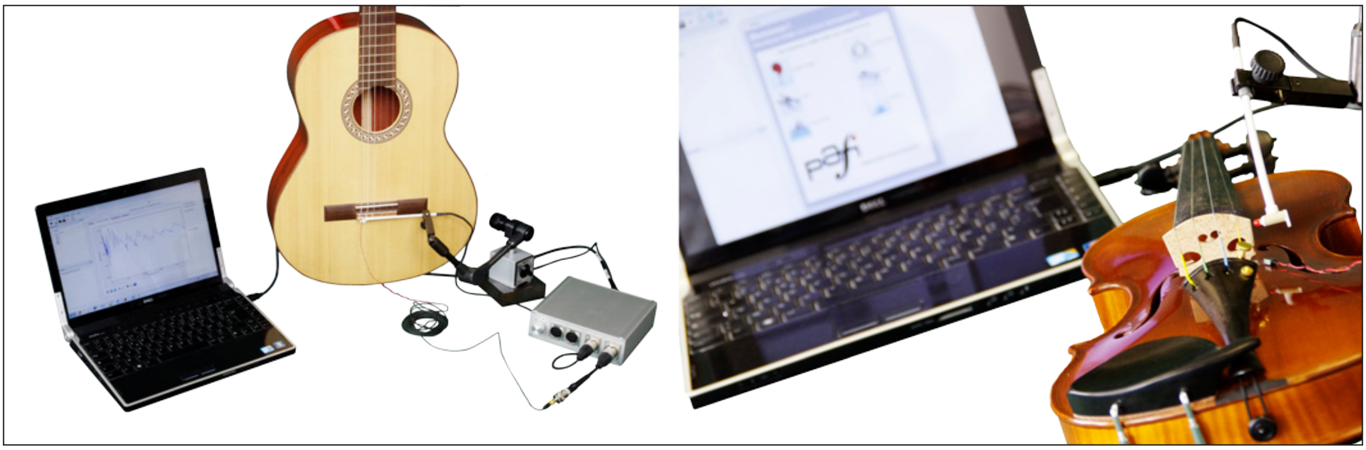


Figure 3. Stringed instrument device prototype (PAFI project): mobility measurement at the bridge. Transfer function measurements at the bridge give important information regarding instrument playing behaviour. It characterizes the overall coupling between the strings and the body by mean of equivalent parameters (see [6], [7], [8], [9]).

implementation of the *Lutherie tools* approach to all the families of instruments: plucked and bowed stringed instruments, wind instruments (woodwind and brass) and, on an exploratory basis, percussion instruments. The objective targeted is the implementation of a hardware and software platform for supporting the design and characterization of musical instruments. The major stake involved is to ensure the compliance of this platform's functionalities with the concrete needs of instrument makers and their seamless integration into their everyday activities.

5.1. Partners and objectives

The project forms part of an ambitious colla borative approach involving four scientific partners (LAUM, Ircam, Télécom ParisTech, and GSII-ESEO[4]), the music professions innovation centre (Itemm), and a group of instrument makers co-ordinated by the UNFI (National Instrument Making Union). The project's strength lies in this combination of skills. What is involved is not just the isolated approach of a single craftsman, but a shared dynamic aimed at the whole of the professional sector.

Currently, top-end instruments are manufactured on a craft industry basis. The difference in assessments made by musicians and instrument-makers from one instrument to the next relates to the aspects of sound emission, expressiveness, ease of playing, and the manufacturing quality: the finish, sturdiness, tuning, etc. This complex concept, referred to using the qualifier "quality", turns out to be tricky to assess. The multitudes of factors which contribute to the development of a successful instrument make it difficult to seek ways to improve it. Indeed, the ideal instrument, as a target to be achieved, does not exist, and does not constitute an unequivocal concept regarding which musicians, instrument-makers and music lovers might agree on. On the other hand, certain aspects of instruments can be deduced by measuring them, such as improving the range, or optimising the accuracy, even though they are linked to judgements based on perception.

In terms of applied research, the project plan is to identify particular indicators which, by using objective characteristics for the instruments, make it possible to categorize and discriminate between instruments using automated methods. For example, these indicators come out of mobility measurements at the bridge of stringed instruments or input impedance measurements of wind instruments. For each of these two instrument families, this work is carried out by students completing their phDs. It also involves seeing in what way the results obtained may be integrated into the craftsmen's various activities: checks on the reproducibility and quality of a manufacturing process, characterization for the purposes of replicating a reference instrument, design support for a new product, selection of materials and sizing, etc.

5.2. Organization and results

Each laboratory specialises in a field, such as in musical acoustics or in instrumentation for the equipment items (force, vibration or acoustic sensors) or in signal processing (traditional Fourier analysis and high-resolution methods). Engineers were hired to ensure the

[4] LAUM, Laboratoire d'acoustique de l'Université du Maine; GSII-ESEO, Groupe Signal Image et Instrumentation de l'École Supérieure d'Électronique de l'Ouest; LTCI-Télécom ParisTech, Laboratoire Traitement et Communication de l'Information de Télécom ParisTech; STMS-Ircam, Sciences et Technologies de la Musique et du Son de l'Institut de coordination Acoustique-Musique.

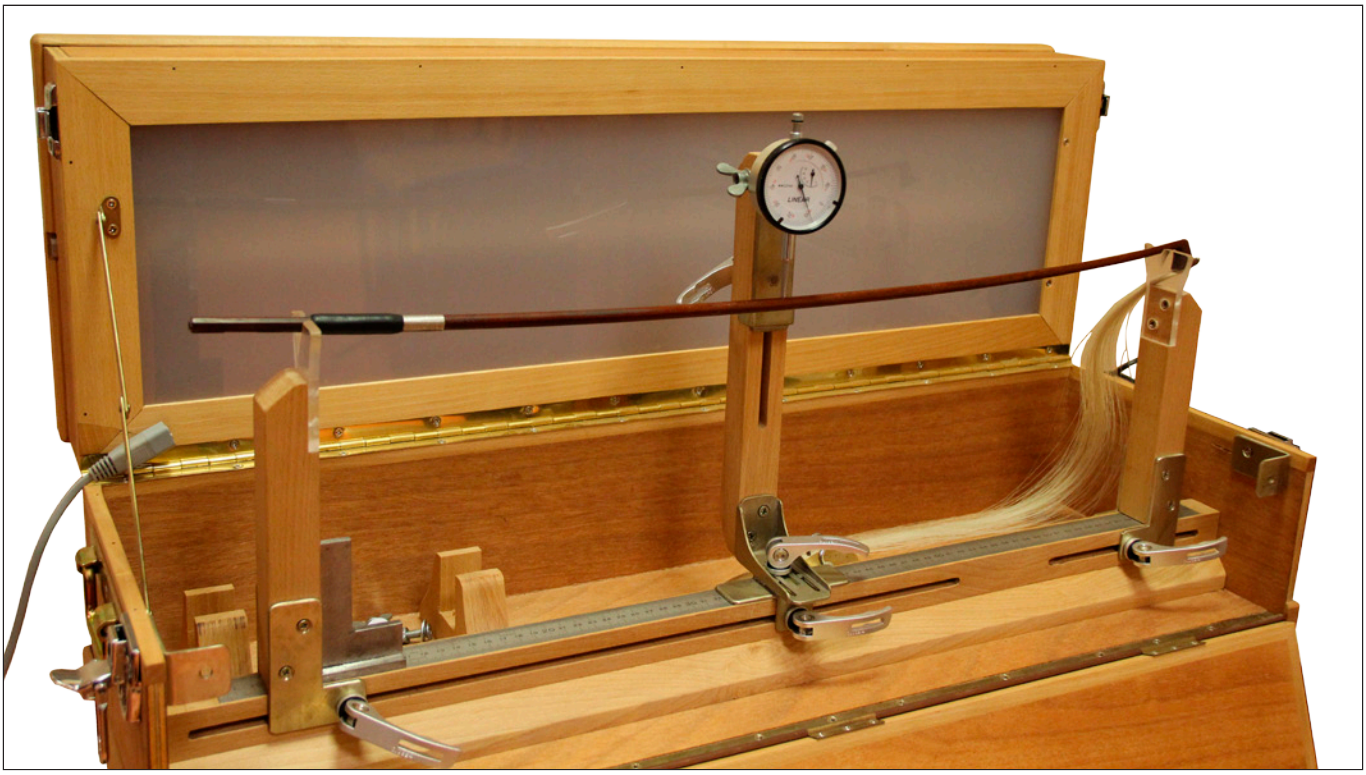


Figure 4. Bow characterization device prototype (PAFI project): mechanical and geometrical parameters measurements.

Figure 5. Wind instrument device prototype (PAFI project): input impedance measurement. The input impedance, measured or calculated, is a quantity that gives important information regarding instrument playing behaviour. It characterizes air column resonances from a given bore geometry (see [13], [14], [15], [16]).

developments and provide support for enterprises. Indeed the project integrated craftsmen right from the time of the design of the products intended for them and the definition of specifications, through to the testing and prototype development. Ten or so "pilot craftsmen" representative of all the various types of instruments were thus selected. Very small instrument making enterprises do not have either the human or financial resources to purchase and adapt laboratory measuring tools. One of the major constraints of the project once again is *to find robust technological solutions at the lowest cost*. It is not the purpose of these tools to find answers to everything, but rather to facilitate innovation within very small enterprises.

The organization of the platform is based on modules: a general module concerning signal analysis and synthesis (for listening educational purposes), and specialized "sector" modules within each family of instruments. The whole lot has to be linked to a

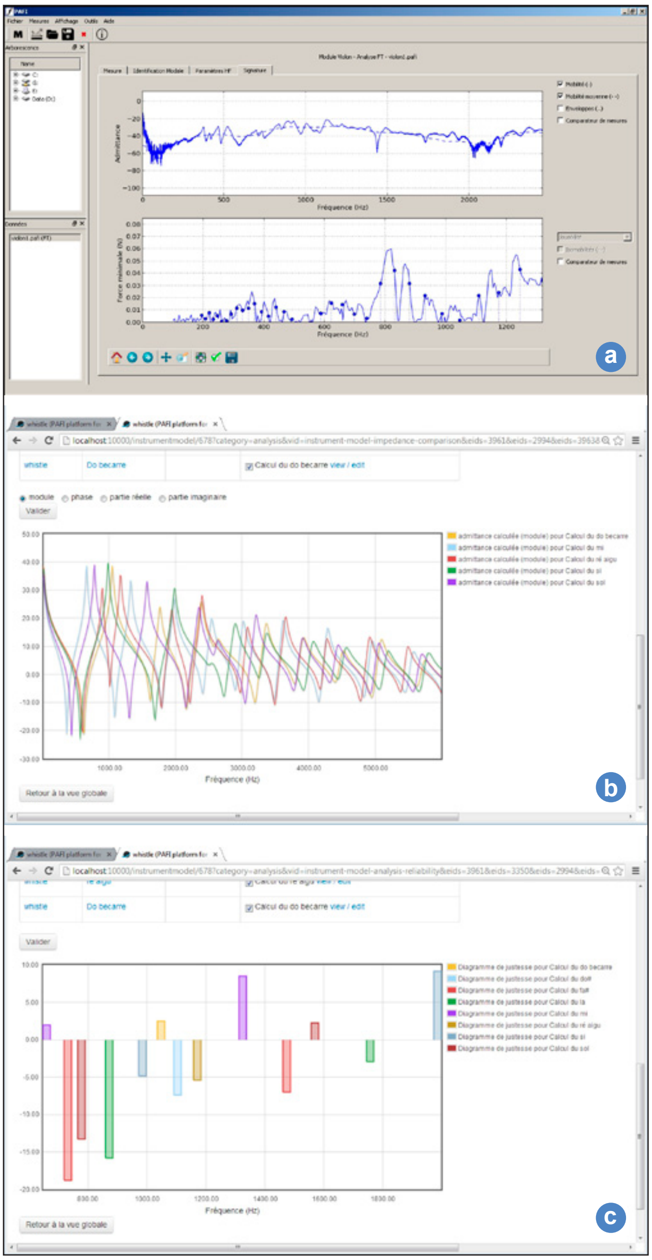


Figure 6. *Analysis and calculating software prototypes (PAFI project).*
a) Stringed instruments software (Python-Qt technologies) – Up: Modulus of measured admittance at the bridge of a violin and its average value (dashed) – Down: Minimal bow force amplitude required to have the Helmholtz motion on the string. It can be seen as a playability criteria (for a given position and bow velocity).
b) Wind instrument software (Python-HTML5 technologies) – Calculation (or measurement) of the input impedance for wood and brass instruments. In this example, several fingering of a whistle are presented as modulus of the input admittance.
c) Calculated whistle prototype intonation diagram deduced from input impedance.

database and a collaborative workspace on the Internet enabling information to be exchanged by the various users of the system. This pooling is intended to make it possible to provide a response to the isolation of craftsmen and create a dynamic that promotes innovation and maintenance of know-how. Simply making such a device available is not enough, and it is foreseen that craftsmen will be trained via training courses organised by the Itemm.

For stringed instruments, a vibrating properties identification methodology of mechanical structures has been developed, whether the object is a musical instruments or not [6], [7], [8], [9]. This methodology has been implemented on several applications: guitars categorization, changes in shape or size of violin bridges analysis, evaluation of different types of guitar bracing, etc. A specific work has been conducted on violin bows in order to ascertain the influence of mechanical and geometrical bow parameters in playing situations [10], [11], [12].

Concerning wind instruments, studies have been logically focused on the exploration of information given by the input impedance in several practical cases [10], [11], [12], [13], [14], [15]. More specifically, the deviations between resonance and playing frequencies have been analyzed. The different methods for calculating the input impedance of an instrument have been compared and put to the test of measurements. With the craftsmen feedback, the scope of the methods have been tested on practical fields such as the influence of pad “resonators” (as the makers named it) on wind instruments keys or the bore reconstruction of a bassoon crook.

The platform is intended to be upgradable and open so that it can be used as the basis for future complementary work. For example, in light of the fact that the scientific results have already been published, it is foreseen that the analysis and calculation software will be published subject to a *free licence* or on an “open source” basis. The aim is to be able to easily integrate new functionalities and to establish a common purpose that may be developed more largely.

6. INTERIM ASSESSMENT, PROSPECTS

This paper presented collective initiatives for technological support to SME involved in instrument making currently under development in France. A lot of work is still to be achieved. Nevertheless, the first results are very encouraging. They enlighten the creation of a virtuous circle of synergy and collaboration between the instrument-making actors and the scientific community.

Upon completion, it is possible that these experiments may serve as a work basis to the general implementation and continuation of this dynamic at an international level. Indeed, in the light of the general typology of handicraft enterprises and the open, upgradeable nature of the work, this prospect may be considered and is even propitious for maintaining the craft industry production of high-quality musical instruments.

REFERENCES

[1] Monsimier J. et al., "*Regards sur la filière instrumentale*", *musique & technique* n°4, Itemm Ed. – ISSN: 1771-3641, pp. 61-108, 2009.

[2] CITES, "*Convention on International Trade in Endangered Species of Wild Fauna and Flora*", http://www.cites.org, 2013.

[3] Doutaut V., "*Facture instrumentale et innovation : enjeux, dispositifs, perspectives*", *musique & technique* n°0, Itemm Ed. – ISSN: 1771-3641, pp. 53-59, 2004.

[4] BIAS, "*Brass Instrument Analysis System*", and VIAS, "*Versatile Instrument Analysis System*", Institut für Wiener Klangstil, http://iwk.mdw.ac.at, 2013.

[5] Gautier F., Doutaut V., Fouilleul J.-M., "*Lutherie tools : projet collaboratif entre ateliers de lutherie et laboratoires*», *musique & technique* n°4, Itemm Ed. – ISSN: 1771-3641, 21-28, 2009.

[6] Élie B., "*Caractérisation vibratoire et acoustique des instruments à cordes : application à l'aide à la facture instrumentale*", PhD thesis, Université du Maine, Le Mans, 2012.

[7] Élie B., Gautier F. and David B., "*Macro parameters describing the mechanical behavior of classical guitars*", Journal of Acoustical Society of America, 132 (6), pp. 4013-4024, 2012.

[8] Élie B., Gautier F. and David B., "*Estimation of mechanical properties of panels based on modal density and mean mobility measurements*", Mechanical Systems and Signal Processing, 40 (2), pp. 628-644, 2013.

[9] Élie B., Gautier F. and David B., "*Acoustic signature of violins based on bridge mobility measurements*", Journal of Acoustical Society of America, submitted.

[10] Ablitzer F., "*Influence des paramètres mécaniques et géométriques sur le comportement statique de l'archet de violon en situation de jeu*", PhD thesis, Université du Maine, 2011.

[11] Ablitzer F., Dalmont J.-P. and Dauchez N., "*Static model of a violin bow: influence of camber and hair tension on mechanical behaviour*", Journal of Acoustical Society of America, 131 (1), pp. 773-782, 2012.

[12] Ablitzer F., Dauchez N. and Dalmont J-P., "*A Predictive Model for the Adjustment of Violin Bows",* Acta Acustica united with Acustica, 98, pp. 640-650, 2012.

[13] Éveno P., "*L'impédance d'entrée pour l'aide à la facture des instruments de musique à vent : mesures, modèles et lien avec les fréquences de jeu*", PhD thesis, Université Pierre et Marie Curie, Paris, 2012.

[14] Éveno P., Dalmont J.-P., Caussé R. and Gilbert J., "*Wave propagation and radiation in a horn: comparisons between models and measurements*", Acta Acustica united with Acustica, 98 (1), pp. 158-165, 2012.

[15] Dalmont J-P., Curtit M. and Fazli Yahaya A., "*On the accuracy of bore reconstruction from input impedance measurements: Application to bassoon crook measurements*", Journal of Acoustical Society of America, 131 (1), pp. 708-714, 2012.

[16] Caussé R., Éveno P., Gilbert J. and Petiot J.-F., "*How can we deduce playing frequencies from measured resonance frequencies for trumpets?*", Journal of Acoustical Society of America, 133, 3548, 2013.

[17] Éveno P., Kieffer B., Caussé R.,Gilbert J. and Petiot J.-F., "*Wave propagation and radiation in a horn: comparisons between models and measurements*", Acta Acustica united with Acustica, submitted.

[18] Éveno P., Curtit M., Dalmont J-P. and Caussé R., "*Influence of pad "resonators" on saxophone admittance*", Stockholm Music Acoustics Conference (SMAC), Stockholm, 2013.

Musician´s perceived timbre and strenght in (too) small rooms

Tor Halmrast
Acoustics/Statsbygg, Norway.
Corresponding author: th@statsbygg.no

PACS: 43.66.Jh, 43.66.Mk, 43.75.Cd

ABSTRACT

Musicians often rehearse in small rooms. This might give problems regarding sound pressure level and timbre (German: klangfarbe). The rehearsal rooms for the Norwegian Armed Forces´ Band North in Harstad were investigated introducing different absorbers (curtain, corner/bass-absorber and wall absorber).

The investigations included calculations/ measurements of room acoustic parameters and recordings/analysis of a short Test Composition for trombone, tuba, Bb-clarinet and Bb-trumpet in the different room settings. Recordings with "in-ear" microphones were used for analysis of timbre and "perceived reverberation". The results indicate that there are two issues that might be more important than plain reverberation time criteria: *1) Room resonances in the bass (tuba and trombone etc.), 2) "Shimmering" for high pitched instruments (clarinet)* A sound source with a given, constant sound power will be reduced some 3-5 dB when a small rehearsal room is modified from "moderate absorption" (curtain) to "well absorbed" (curtain, corner absorber and some wall absorbers. Musicians will compensate unconsciously, so effective reduction will in practice be some 1-2 dB less.

1. TEST COMPOSTION

A one minute test composition was prepared for the investigation, see App. A. Fig. 1 shows the trombone version. The version for trumpet is simply an octave transposition, but the versions for tuba and clarinet were further modified with extra octave transpositions of sections in order to fill the tonal range (ambitus) of each instruments (clarinet extra ***8va*** for some phrases and some tuba phrases additional ***8va*** basso). The total "score" of the Test Composition is shown in Appendix A, also for additional instruments for further investigations. The instrumental recordings were done with a Sennheiser "In-Ear"-Microphone and simultaneously with a calibrated omni microphone typically 1.5 m and 45^0 from the bell of the instrument. All recordings were done in "wav", 16 bits, 44.1 kHz. Each instrument was recorded solo, in strict tempo MM q = 120 using a silent, flashing metronome.

Figure 1. Test Composition. Trombone.

2. THE REHEARSAL ROOM

The rehearsal room has dimensions: L = 4.33m, W = 2.14m, and H = 2.28m. A calculation of room resonances is given in Appendix B.

Figure 2. Room with absorbing curtain only (Non-dampened room).

The only absorbing material is a flexible curtain which was present in all the musical tests. In order to eliminate any "false reverberation", there was no piano in the room. The extra absorbers introduced were a corner/"bass" absorber in form of a roll of Rockwool, and 4 Rockfon Cosmos wall absorbers (thickness 100mm) (see fig. 3)

Figure 3. Corner/"bass"-absorber and Cosmos wall absorber (Dampened room).

All walls and the ceiling are in gypsum boards, and the floor is a light, floating floor on mineral wool. A rough Odeon (Sketch) model of the room is shown in fig. 4.

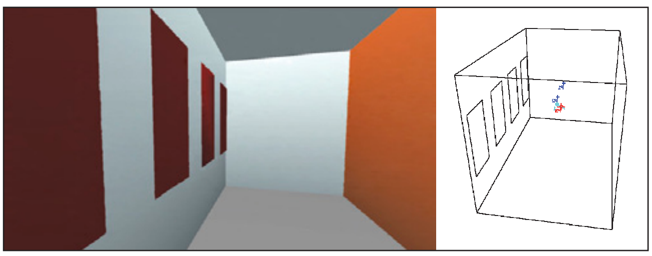


Figure. 4 Odeon model of the room (Dark = absorber).

3. REVERBERATION TIMES

3.1. Calculated/Simulated Reverberation Times

An overall Odeon Reverberation Time calculation gives that the room with plain gypsum walls (no curtain) will have a too long reverberation time (see fig.5). In daily use the only absorber in the room was a curtain on one of the sidewalls, one chair and a bench. This situation is referred to as *"non-dampened".* The calculated reverberation time for this situation also gives rather high values. (see Fig. 5)

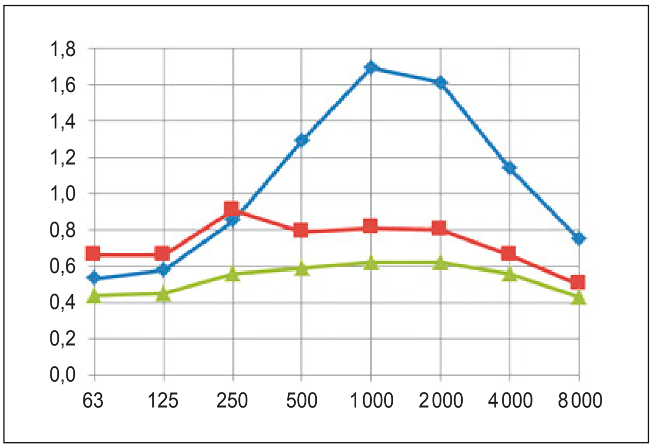


Figure 5. Results from Odeon simulation of rehearsal room.
Upper: Gypsum on all walls.
Middle: Curtain on one sidewall.
Lower: Curtain and wall absorbers on the other side wall.

These preliminary room simulations might give the impression that the reverberation time is sufficiently low in the bass due to the bass absorption of the gypsum walls. We shall see that we do have problems with room resonances, and need to introduce a bass-/corner-absorber.

3.2. Measured Reverberation Times

The reverberation times and Schroeder curves with/without the flexible curtain are shown in fig. 6.

For the situation with flexible curtain and corner/"bass-absorber, we get the following reverberation time (fig. 7).

We see that the corner absorber gives nice reduction for the bass (80-250 Hz). The moderate rise in reverberation time for mid-frequencies (1 kHz - 4 kHz) when introducing the corner absorber is somewhat strange, but might, to a certain degree, be explained by the fact this "low-budget" corner absorber is covered with plastic, which gives some reflections closer to the measuring microphone. The most important is the nice reduction for the bass.

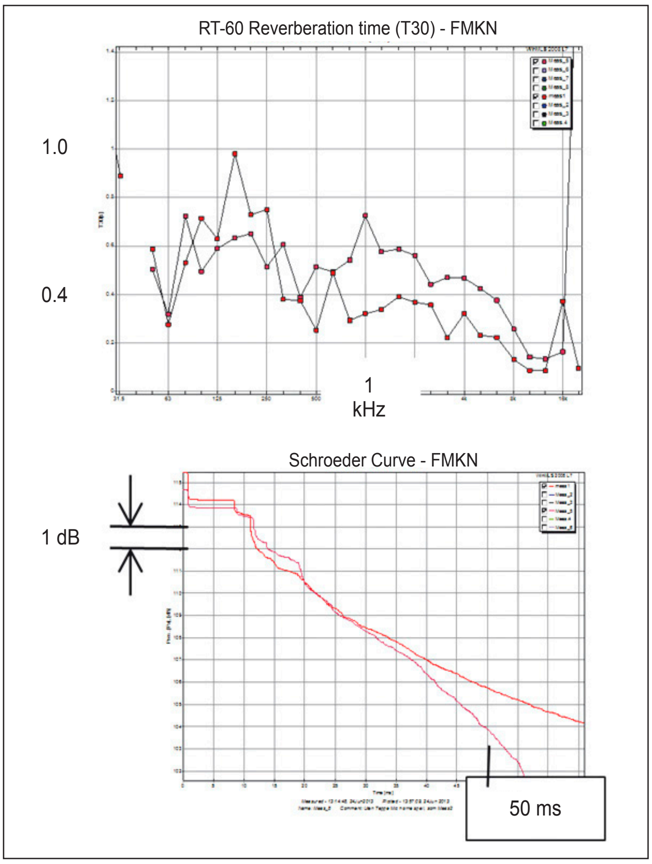


Figure 6. *Reverberation Time and Schroeder curves with/without flexible curtain.*

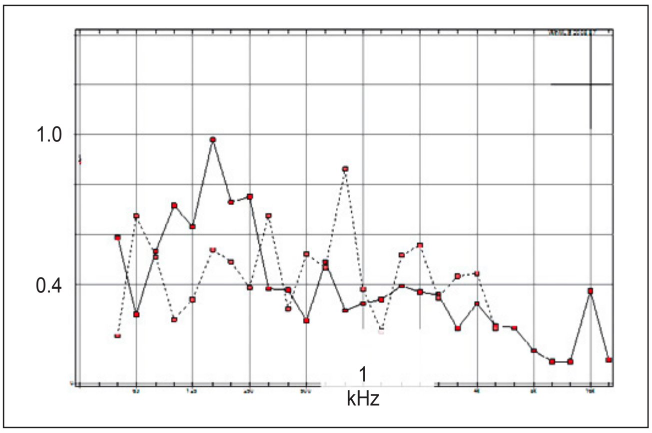


Figure 7. *Reverberation time with/without corner absorber (in addition to flexible curtain).*

When introducing the Cosmos wall absorbers in addition, we get the reverberation time shown in fig. 8. (for two receiver positions).

We see the known fact that measuring reverberation times in small rooms is highly sensitive to microphone position, due to the resonances, especially for the lower frequencies (see Appendix B). Usually one takes the average of several measurement positions to get a statistical value, but that will "hide" the observed problem of room resonances in small rooms for music and their influence on the perceived timbre, which is a main issue for this paper.

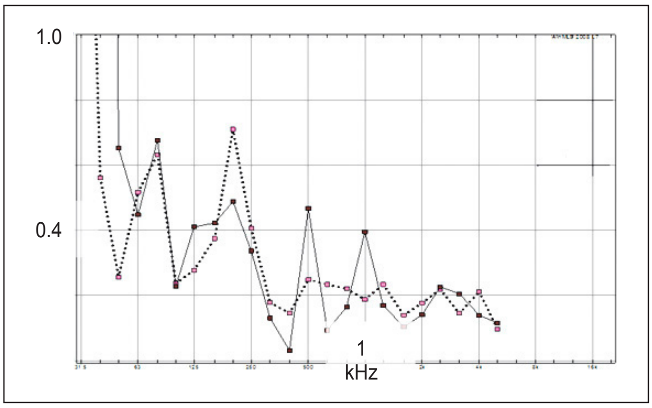


Figure 8. *Reverberation time with all absorbers (2 measurement points for the same situation).*

A simplified overview of the measured reverberation time for the different settings of absorbers is given in fig. 9.

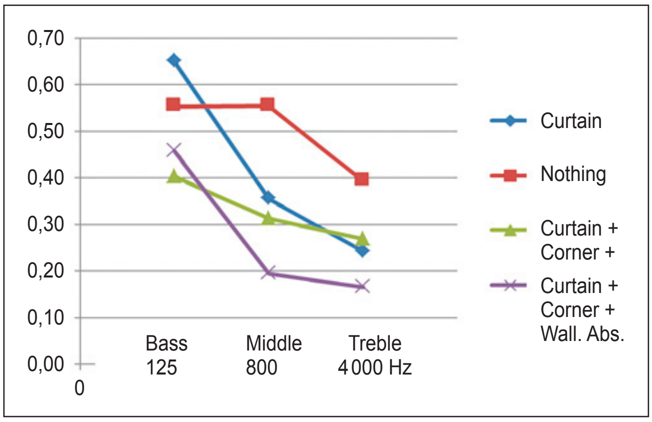


Figure 9. *Overview of Reverberation Times (Tm: Nothing: 0.5, Curtain: 0.42, Curtain+Corner: 0.33, All: 0.27).*

We see that the curtain reduces the reverberation time for middle frequencies from some 0.55 s to some 0.35 s, and the four wall absorbers reduces the reverberation further down to some 0.2 s for these middle frequencies. For the bass frequencies the corner absorber is the most important. The results are somewhat unclear due to the resonances and because we included just a few measuring positions, but the reduction due to the corner absorber is some 0.2 s in the bass. The absorption coefficient for such a corner absorber is not given in text books or product information, but by a very rough estimation from the reductions of reverberation times in fig. 7, the corresponding absorption area of such a simple corner absorber, just a roll of Rockwool, is some 4m^{2}Sabine for the frequency region 80-250 Hz (see Appendix E).

3.3. Musician´s Perceived Reverberation

As stated in [2] it is possible to use your hand-claps and tongue drops (clicks) recorded with microphones in your own ear to judge your perceived reverberation in a room for middle frequencies. When evaluating the reverberation times from such recordings, we must eliminate the strong direct sound by taking the calculations from some -20 to -35 dB (instead of from -5 to -35 dB as for standardised measurements with longer distances between source and receiver, as in ISO 3382-1 and -2).

Figure 10 shows the Schroeder curves and reverberation times the for tongue drops in the "dampened" situation (red) and in the "non-dampened" situation (blue).

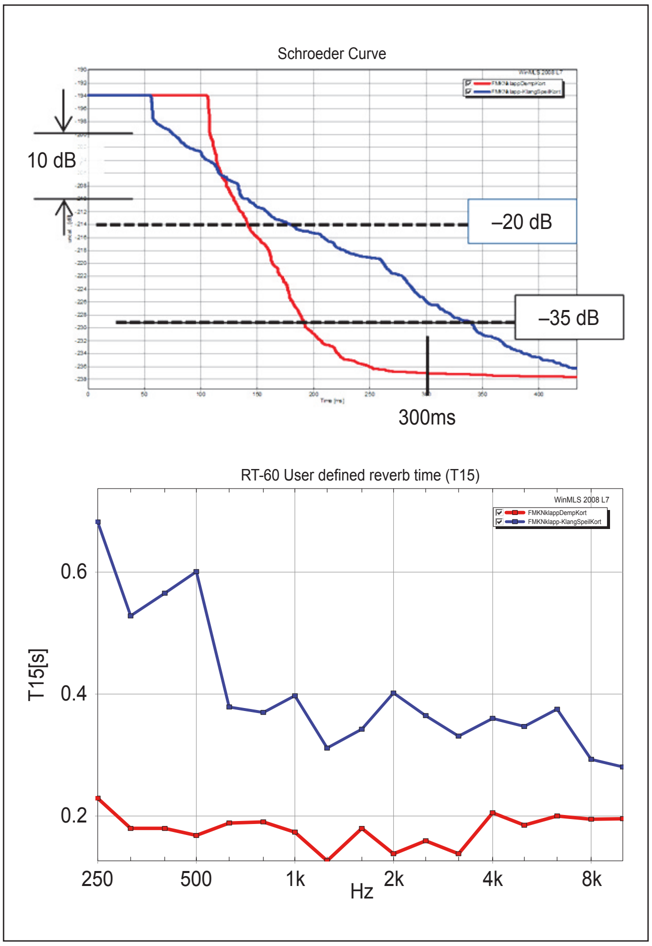


Figure 10. Schroeder curves and Reverberation times for In-Ear measurements of own tongue drops. (User defined T15, taken from -20 dB to -35 dB). Blue: Non-Dampened room. Red: Dampened room.

We see that the dampened room is perceived ("in-ear") as having an even shorter reverberation time than what was measured by standardised methods in 3.2.

4. SOUND PRESSURE LEVELS

4.1. Measurements with loudspeaker

The measured reduction of sound pressure level when introducing the different absorbers is shown in fig. 11.

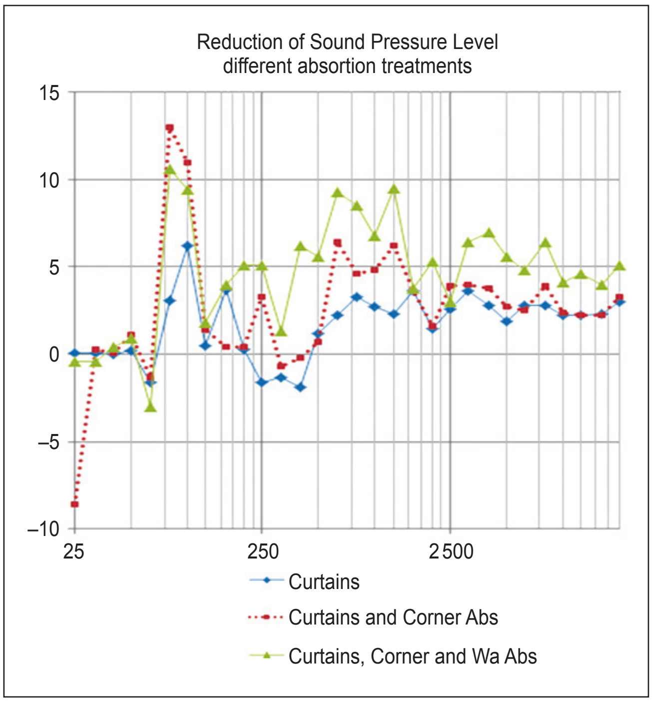


Figure 11. Reduction of Sound pressure level by introducing different absorption treatments (1/3 octave) Curtains only (mean = 1.7 dB) Curtains and Corner Absorbers (mean = 2.6 dB) Curtains, Corner and Wall Absorbers (mean = 4.7 dB).

The measurements more or less follow the theoretical 3 dB reduction in level when dividing the reverberation time by two. Such a calculation is of course only correct for diffuse sound fields. Our small rehearsal room is far from diffuse due to dominating room resonances and small size, but the result is interesting.

For the middle frequencies (315 - 2000 Hz) we have the following mean values for the reduction of sound pressure levels*: Curtain: 1.5 dB, Curtain + Corner: 3 dB and Curtain+Corner+Wall: 6.5 dB),* so that the change between having just curtains ("non-dampened") and all absorbers ("dampened") is 5 dB.

4.2. Measurements of musicians playing the Test Composition

Unfortunately it is not possible to calibrate the Sennheiser "in-ear"-microphones with a pistophone. Therefore, calibrated sound pressure levels were measured simultaneously with a calibrated omni directional microphone positioned app. 1.5 m from the instruments, 45° to the side of the main direction of the "bell". First we will look at these "in the room" recordings in the "non-dampened" rehearsal room (just curtains), compared to the "dampened" situation with all absorbers (corner and wall absorbers in addition). Later we will look more into details of the "in-ear"-recordings, (uncalibrated, but recorded at equal input level, so that comparisons between them are possible).

4.3. Calibrated, "in the room" measurements of sound pressure levels from musicians

A histogram of dBA_{fast} versus time for the whole Test Composition (trombone) is shown in fig. 12, with/without curtains.

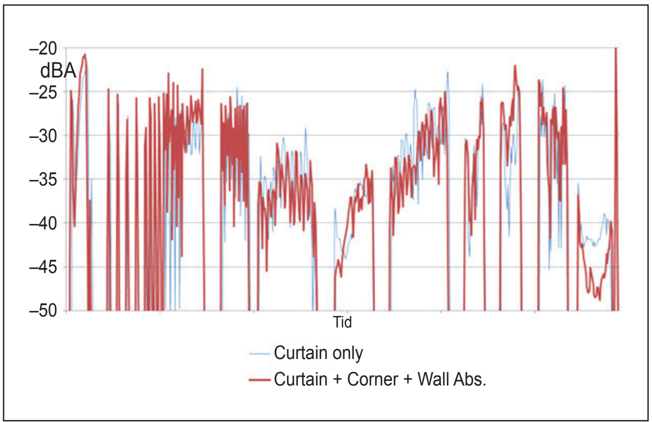


Figure 12. *Histogram of dBA/time. With/without curtains.*

The middle part of the piece might indicate a 2-3 dB reduction with the curtain, but we find that in this representation, the difference between the levels of the recordings with different room damping for the trombone is not very clear.

We will now look at the dBA_{fast} levels for the 10 s forte, ***f***, section of the test composition (marked in red on top of each curve in fig. 13) for both the "dampened" (all absorbers) and "non-dampened" (just curtain) situations, for each instrument. Shown is also the energy based equivalent levels, Leq for the whole test composition (marked with blue circles).

Apart from the clarinet, we see the following reductions in Lp for the 10 s forte, ***f***, section (red circles):

- *Tuba: 102 – 100 = 2 dBA*
- *Trumpet: 103 – 99 = 4 dBA*
- *Trombone: 101 – 97 = 3 dBA*

For Leq (dBA) measured "in the room" for the duration of the whole test composition (blue) we find:

- *Tuba: 100.5 – 97.2 = 3.3 dBA*
- *Trumpet: 94.9 – 90.6 = 4.3 dBA*
- *Trombone: 95.5 – 92.0 = 3.5 dBA*

For clarinet seems to "over"-adjust to the damped acoustics, and play stronger. The Leq is actually higher in the dampened room than in the more reverberant! This might be "personal", and is discussed later.

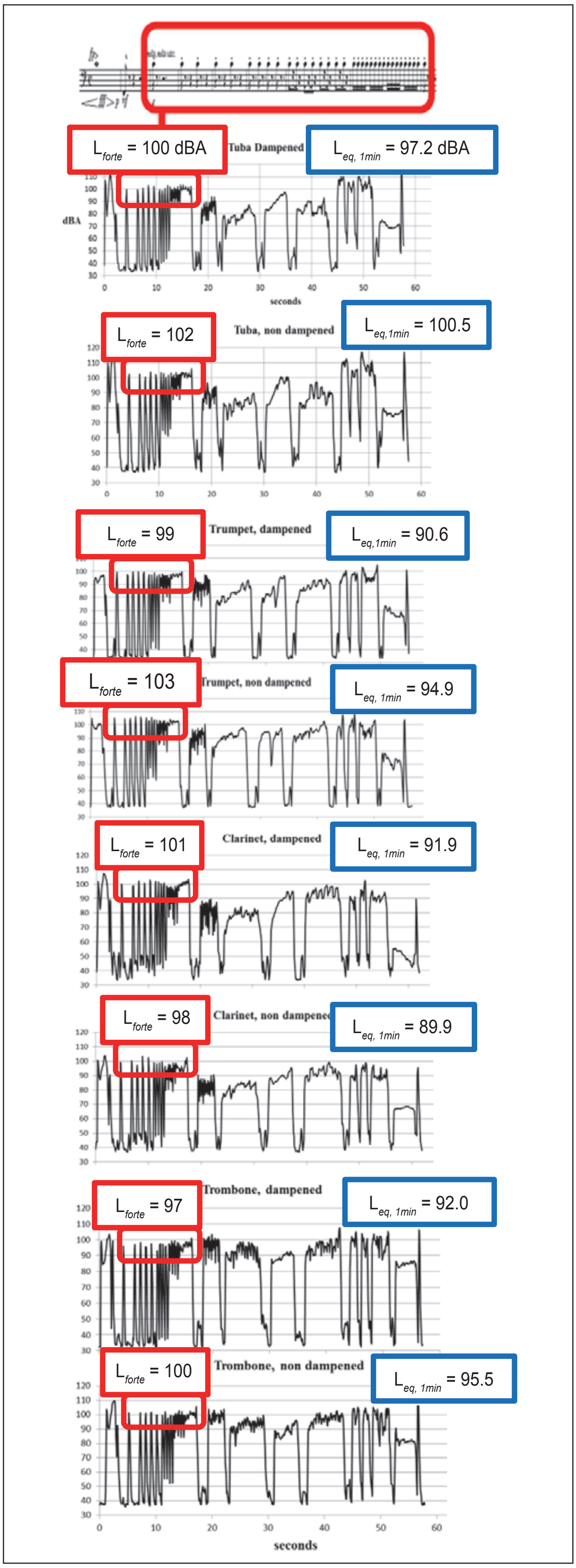


Figure 13. *Histogram dBA_{fast} versus time for instruments. Test Composition in Dampened/Non-Dampened room. Red squares indicate forte,* ***f****, section.*

Conclusion, Sound Pressure Level: The measured reduction in sound pressure level at the musician´s ear when introducing the extra absorbers is 1-2 dB less than the reduction measured with a constant loudspeaker source. This means that the musicians (not only the clarinet) compensate for the reduced “answer” from the room by playing 1-2 dB stronger.

4.4. *Comparisons with theoretical studies*

Mayer [1] gives information on typical Sound Power Levels, Lw. The following table shows an adaption.

Instrument	*(ppp)*	*pp-p*	*p-mf*	*f*	*ff-(fff)*
Tuba	76	93	<<<	**106**	112
Trumpet	77	89	<<<	**101**	111
Clarinet	55	75	<<<	**94**	107
Trombone	73	93	<<<	**101**	113

For our further calculations, we will use the sound power levels for ***f*** in this table and compare with the measured results from 4.3.From general acoustic theory we have the following equation between sound power (Lw) for a sound source and sound pressure level (Lp) in a room with Volume V and Reverberation time T: [1]

——— ——— *(1)*

where Q is the directivity factor for the source. For a rough estimate one might be tempted to assume a diffuse soundfield and discard the direct sound by dropping the first part of the parenthesis. This would of course not be quite correct in such a small room, so in the following table we have calculated the sound pressure levels both: a) without this first part of the parenthesis, (first numbers in black in the table) and b) with the first part of the parenthesis (last numbers in black in the table below). A Q-factor of 1 is chosen (as for a point source)[2]. Our room Volume is 2.4 x 4.33 x 2.28 = 23.7 m^3, so for the different reverberation times, we get the following calculated sound pressure levels (Lp) for *forte*, ***f***, compared the measured ones from 4.2 (in red and parentheses):

[1] Sound pressure level can be taken using the parameter Strength, G [dB], but we did not have calibrated measurement equipment available.

[2] The measuring position r = 1.5 m and 45^0 from the direction of the bell might give a Q somewhat lower than 1.

	T = 0.5 s	T = 0.35 s	T = 0.2 s
Tuba	103.2-103.6 **(102)**	101.7-102.3	99.2-100.2 **(100)**
Trumpet	98.2-98.6 **(103)**	96.7-97.3	94.2-95.2 **(99)**
Clarinet	91.2-91.6 **(98)**	89.7-90.3	87.2-88.2 **(101)**
Trombone	98.2-98.6 **(100)**	96.7-97.3	94.2 -95.2 **(97)**

The measured values are in very god agreement with theory for tuba and trombone, but some 4 dB higher for the trumpet. The clarinet is 8-10 dB stronger than calculated; see the comments on clarinet in 4.3 and 4.5. The “strange” behaviour of the clarinet is probably in order to avoid the “shimmering” timbre, se chapter 5.

4.5. *“In-Ear-Measurements” of Test Composition*

Detailed analysis was performed also on the “in-ear”-recordings (which were done simultaneously with the “in room” recordings discussed in 4.3). As mentioned, these measurements are not calibrated dB SPL, but all these measurements were performed with the same settings, so comparisons between them are possible. We will first look into the recordings of the clarinet, because this instrument showed somewhat strange result in the previous chapter. The “in ear”-histograms for the “dampened room” and the “non-dampened room” are shown in fig. 14.

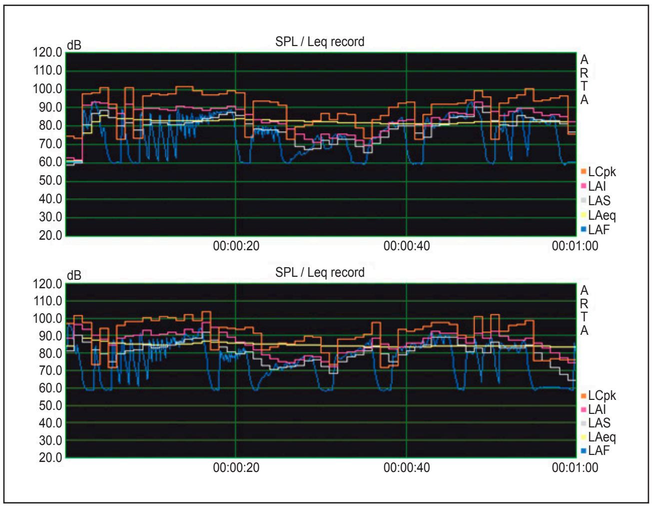


Figure 14. *Histogram In-Ear-Recordings of the Test Composition. Upper: Dampened Room, Lower: Non-Dampened room Clarinet (non-calibrated).*

The following table gives the main results: (not calibrated)

CLARINET	LAeqT	LCpeakmax	Sone	Phon
Whole Piece				
Dampened	81.9	101.7		
Un-Dampened	83.3	103.9		
10 s Intro				
Dampened			42.3	94.0
Un-Dampened			50.4	96.5

We see that these “in-ear”-measurements” give some 1.5 – 2 dB reductions for the clarinet when the room is changed from “non-dampened” (just curtains) to “dampened” (curtains, corner absorber, wall absorbers), so these “in-ear-recordings” indicate that the additional damping with corner and wall absorbers give a perceived “in-ear” reduction also for the clarinet. The Phone value is also reduced some 2.5 when dampening the room.

For the Trumpet, we get the Histograms shown in fig. 15

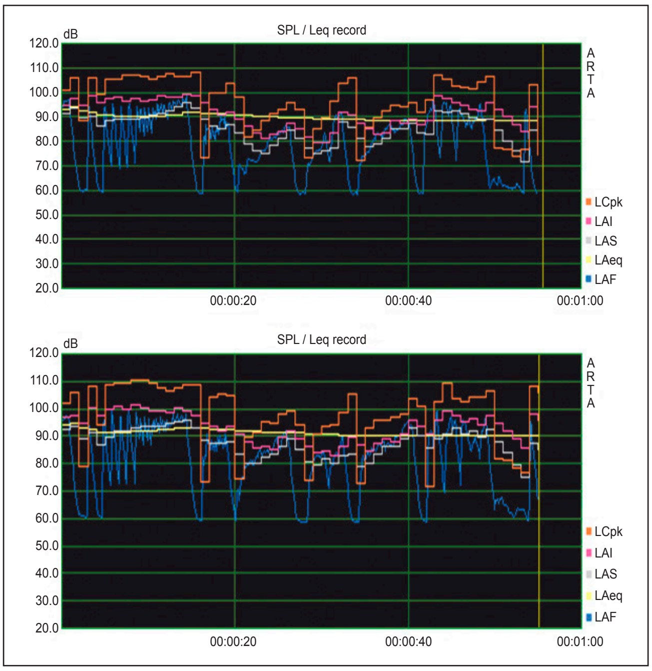


Figure 15. *Histogram In-Ear-Recordings of the Test Composition. Upper: Dampened Room. Lower: Non-Dampened room Trumpet.*

The main results for trumpet: (not calibrated)

TRUMPET	LAeqT	LCpeakmax	Sone	Phon
Whole Piece				
Dampened	88.4	108.5		
Un-Dampened	89.9	105.2		
10 s *forte, f* Intro				
Dampened			71.8	101.7
Un-Dampened			90.2	105.0

We see that Leq for the trumpet is reduced some 1.5 dB when dampening the room[3]. The reduction in Phone for the whole piece was some 3.2 dB, which is about 1 dB less than the reduction for the “in room” measurements in 4.3. However, for this recording of the trumpet, the max level during the period was actually higher when the room was dampened (see table above). This is due to one single ***ff***-note played strong, and is not statistically relevant, but shows that dampening a room is not, by itself, necessarily a security for lower sound pressure levels for short, strong notes.

4.6. Sound pressure levels when playing pia-pianissimo

All discussion so far has been for ***f*** (or ***ff*** and ***fff***). Analysis for a part of the Test Composition that calls for “*very smoothly* ***pp***” is shown the in figure 16. (The fact that the played tones are not very “steady” in strength will not be discussed further. We will look at the mean value for the “red” sections).

From fig. 16 we see that for the brass instruments the sound pressure levels for the ***pp*** section are reduced by 2-4 dB when dampening the room. This is almost the same as for the 10 s section of the composition played ***f*** (1 dB lower than in 4.3). So, the conclusion for brass playing ***pp*** is about the same as for playing ***f***. This means that: **Musicians (brass) use the same dynamic range both for dampened and non-dampened room,** (when asked to following the indications in the score).

For the clarinet we see a 10-11 dB reduction from “non-dampened” to “dampened”, which again might be “personal” or an example of the fact that a clarinet is more easily played soft than brass instruments and thus might use a larger dynamic range.

We also need to consider that the rather high background noise in the room (see App. D) might influence when playing ***pp***).

4.7. Loudness when musicians face different surfaces

It is commonly believed that one should not rehearse playing directly into a reflecting wall. For the non-dampened situation, the trombone was recorded playing a) Towards the curtain, b) Towards the reflecting wall (with gypsum and mirrors) and c) Towards the door (the

[3] We should notice that the Sone levels, which might be a better parameter for perceived loudness, show a greater reduction.

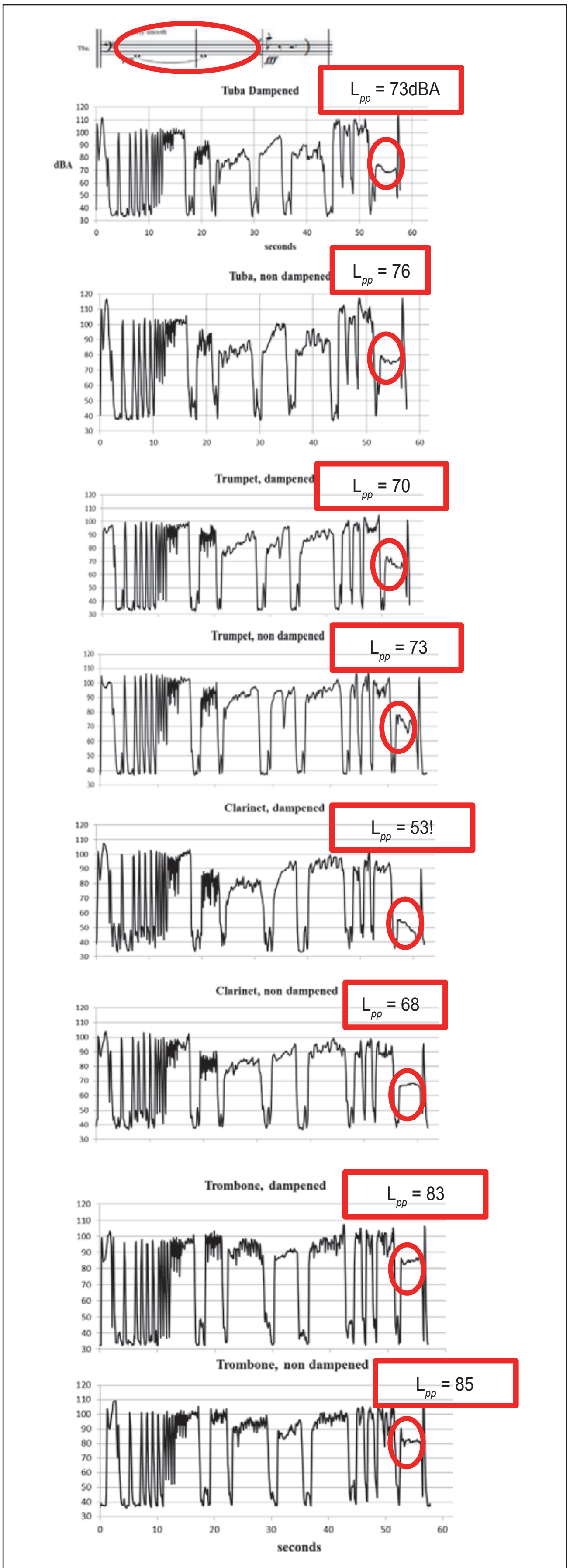


Figure 16. *Histogram dBA versus time for instruments.*

length of the room). The differences of sound pressure levels as mean values for the whole test composition were surprisingly low (within +/- 0.7 dB), and thus not significant. Actually the measurement towards the curtain was the strongest, followed by the one facing the door. For changes in timbre for the different directions, see 5.2.

4.8. Perceived "In-Ear" changes in Strength (G)

Our measuring equipment did not include a calibrated sound source for measuring Strength, G, but we did measure the differences of uncalibrated G with tongue drops/clicks recorded in the musician´s own ear as signal.

Figure 17 gives the results from analysing G from "in-ear recording" of own tongue clicks in the different settings for the rehearsal room.

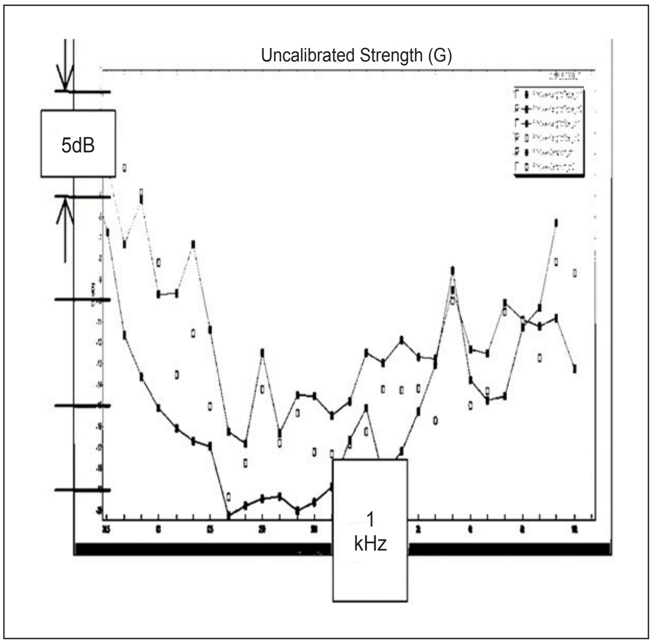


Figure 17. *Musician's perceived reduction of uncalibrated G (strength) of In-Ear recording of own tongue clicks. Upper: Non Damped room (just curtain) Lower: Damped room (curtain, corner, wall).*

We see some 3-5 dB reduction of Perceived Strength (G) when introducing all absorbers. (The measurement method using tongue drops does not give sufficient signal to noise ratio for the higher frequencies, so the measurements for this frequency range (and the low bass) will not be discussed further).

5. SPECTRUM "KLANGFARBE"/TIMBRE

5.1. "In-Ear"-measurements of Timbre

The "in-ear"-recordings were analysed in spectrograms/ sonograms. (see fig. 18). Notice that length of short notes is somewhat longer in the non-damped settings (lower part of fig. 18).

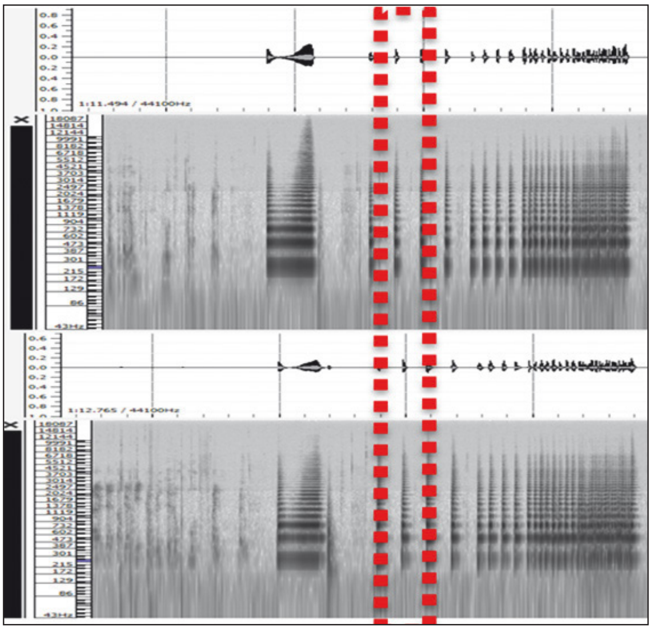


Figure 18. *Clarinet. First part of Composition. Upper: Dampened Room. Lower: Non Damped room. Increased length of notes when longer reverberation. (logarithmic frequency scale).*

Changes in timbre were clearly heard between different settings of absorbers. However, these perceived changes are not clearly seen using common settings for overall sonograms for the tuba. (see fig. 19).

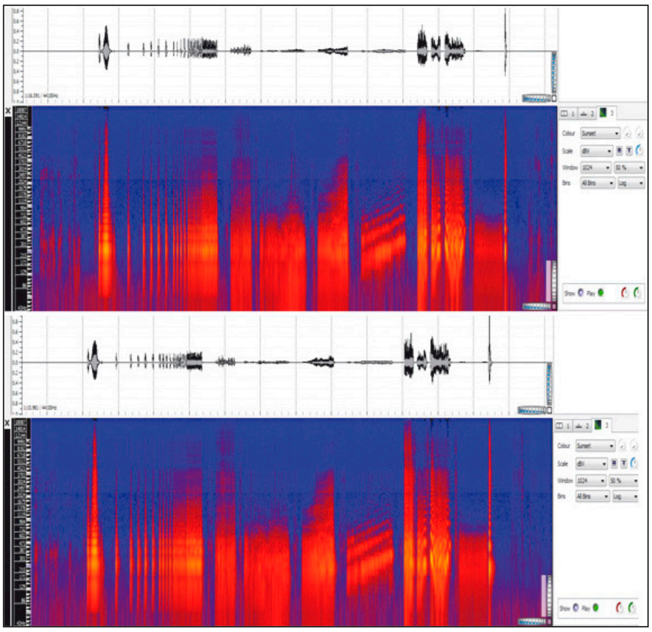


Figure 19. *Sonogram of Tuba "in ear"-recording. Upper: Damped room. Lower: Non-Damped Room. (log frequency scale).*

A more precise frequency analysis of the trumpet, however, shows some 5 dB increase for frequencies around 2000 Hz, and also some 13 dB increase for the peak slightly over 500 Hz. (see fig. 20 and 21 for trumpet).

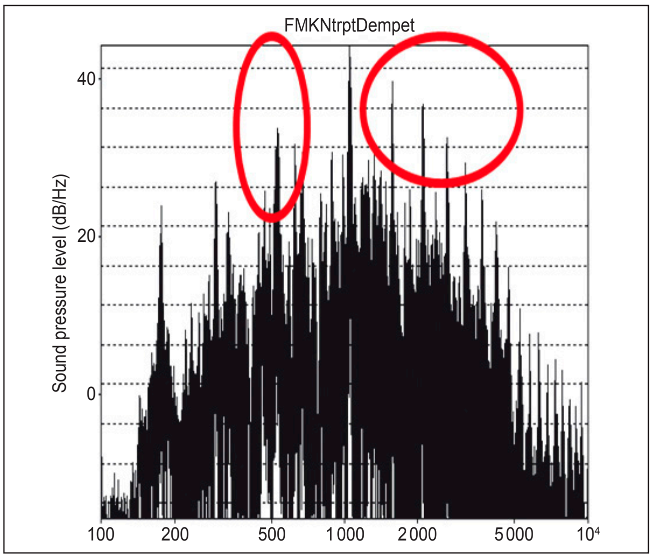


Figure 20. *Frequency analysis of the whole piece. Trumpet. Dampened room. Logarithmic scale.*

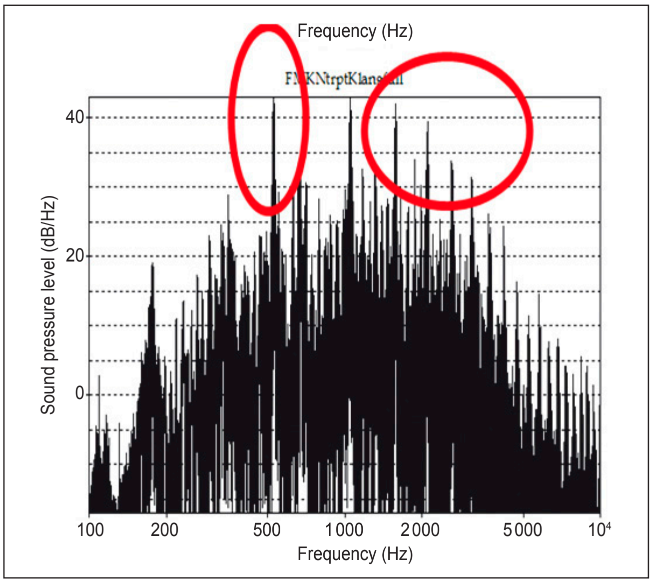


Figure 21. *Frequency analysis of the whole piece. Trumpet. Non-dampened room. Logarithmic scale.*

Also for the clarinet, the perceived changes in timbre when playing in the non-dampened room, was clearly heard, especially for the higher register, but not easily detected using common settings for sonograms. (see fig. 22).

Even a frequency analysis of the whole Test Composition does not show very clear difference between "non-dampened" and "dampened" (se fig. 23), but there seems to be a reduction at 3000 Hz.

The clearly perceived changes in timbre for the clarinet were mainly heard in the high register at forte, ***f***. Therefore we will again examine the first 10 s forte-section of the piece (after the first note) more closely (the same section as investigated in fig. 13, but now for clarinet two octaves higher, sounding high C "natura", around 1 kHz).

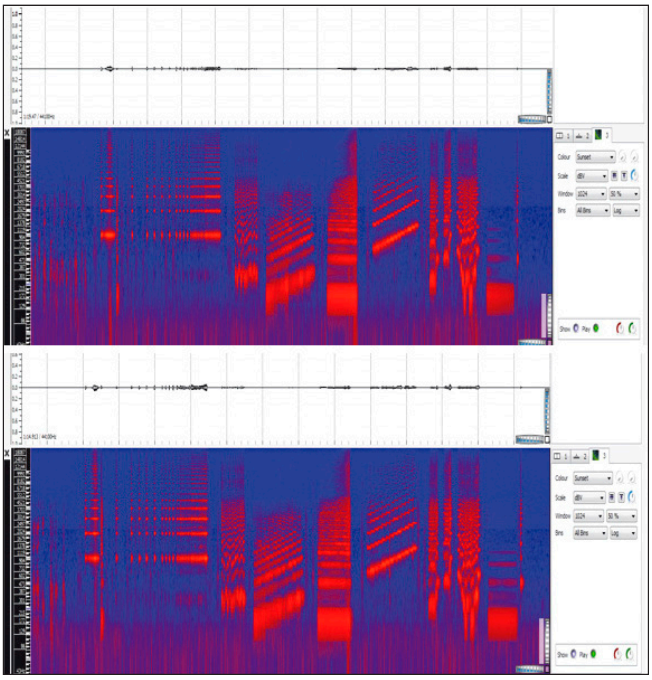


Figure 22. *Sonogram of Clarinet "in ear"-recording. Upper: Damped room. Lower: Non-Damped Room. (log frequency scale).*

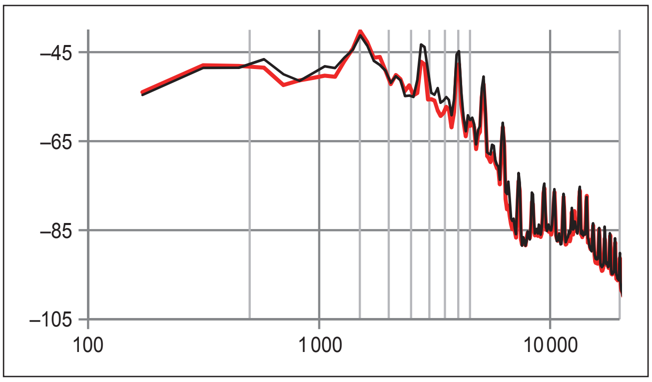


Figure 23. *Frequency analysis of whole Composition. Clarinet. Red = Dampened room. Black = Non Dampened.*

If we take frequency analysis of these 10 seconds of the piece, we see clear changes in timbre between damped and non-damped room, especially for the second, third and fourth partials which are increased by some 5 dB. (see fig. 24). Even the fifth partial is clearly increased. This gives the unpleasant sharpness for this high register in the "non-dampened" room.

This is even clearer when using a logarithmic frequency scale (see Fig. 25).

This increase in higher frequencies can be seen also when analysing the Spectral Centroid (see fig. 26).

Analysis in Pure Data (Pd) (see Appendix C) gives that the peak spectral centroid for the whole, short 10 s section is raised from 3617 Hz to 3930 Hz for the "non-dampened" room compared to the "dampened" room for the clarinet. Similar analysis for the trumpet gave a rise in Spectral Centroid from 3132 Hz for the dampened room, to 4281 for the non-dampened room.

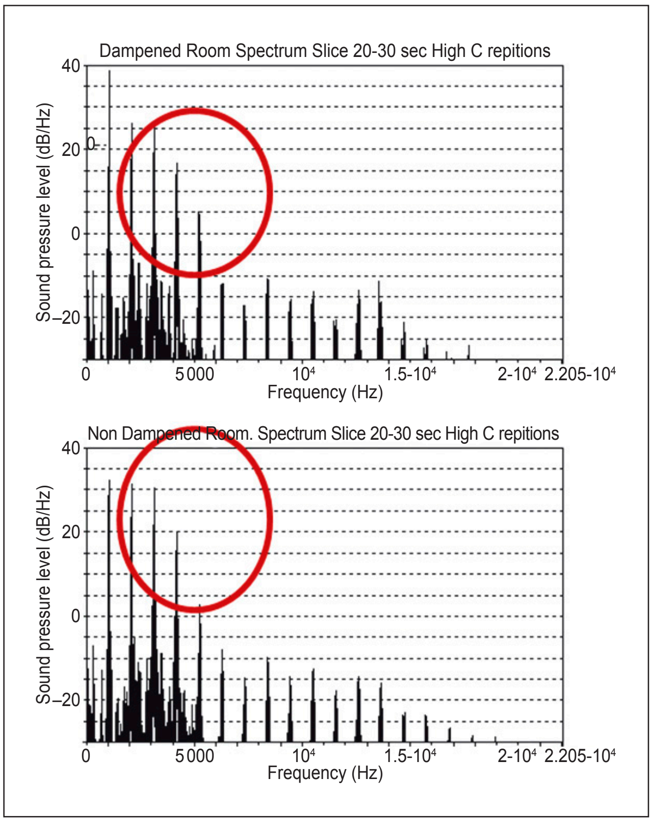


Figure 24. *Freq. analysis of 10 s of clarinet playing high C natura (app. 1kHz). Upper: Dampened Room. Lower: Non-Dampened Linear freq. scale.*

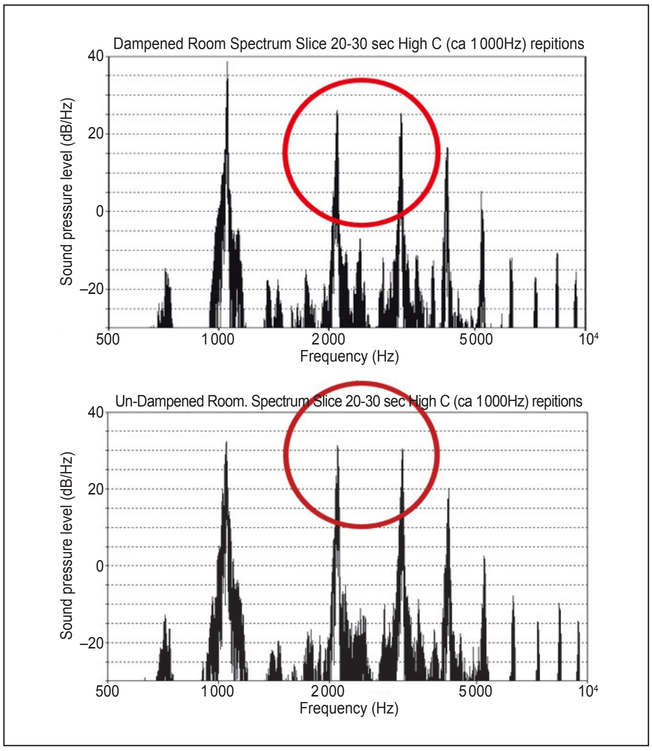


Figure 25. *Freq. analysis of 10 s of clarinet playing high C natura (app. 1kHz). Upper: Dampened Room. Lower: Non-Dampened Logarithmic frequency scale.*

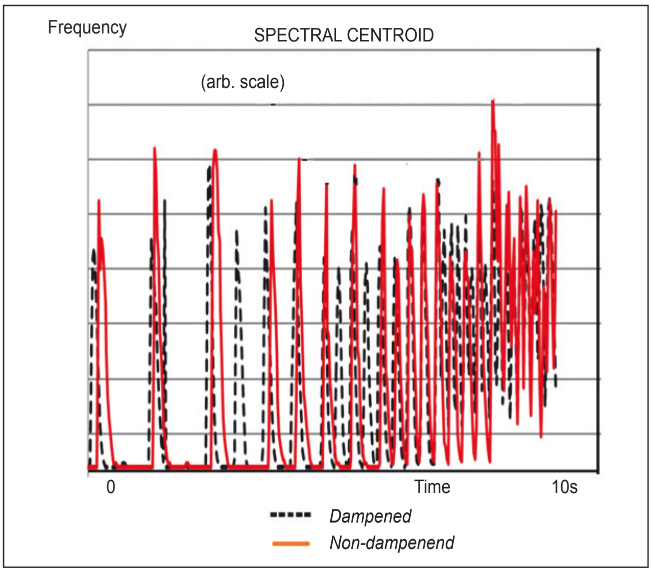


Figure 26. Spectral Centroid versus time of the 10 sec of high pitched clarinet, 10 s of short notes just after the beginning of the test piece.

"By ear" this high frequency "shimmering" was more annoying for the clarinet than for the trumpet. This is of course mainly because the clarinet plays this section one octave higher than the trumpet[4], but perhaps also because the clarinet produces only odd numbered harmonics, so that the partials are more spaced.

The unpleasant sharpness for 2-4 kHz is in the frequency region where the human ear is most sensitive and the kind of sounds we should be most aware of regarding the possibility of hearing loss.

5.2. *Timbre when musician face different surfaces*

For the non-dampened room the trombone was recorded playing the Test Composition in different directions: a) Facing towards the curtain sidewall, b) Facing the reflecting sidewall (mirror and gypsum) and c) Facing the short wall with door towards the corridor.

In 4.7 it is shown that we, surprisingly, did not get significant differences in sound pressure level for the different directions. Also for the frequency analysis in fig. 27, the differences are not large. Towards the curtain gives somewhat less for the 150-200 Hz region. Facing the door gives some dB more in the mid-frequencies. This is surprising because this is the direction with the longest distance to a surface in the direction of the bell. The reason why this direction is the "strongest" might be that it is the direction that gives the trombone player the most freedom, and it his favourite direction when rehearsing in such small room.

[4] The situation might be the opposite for a big band lead trumpet playing *8va*.

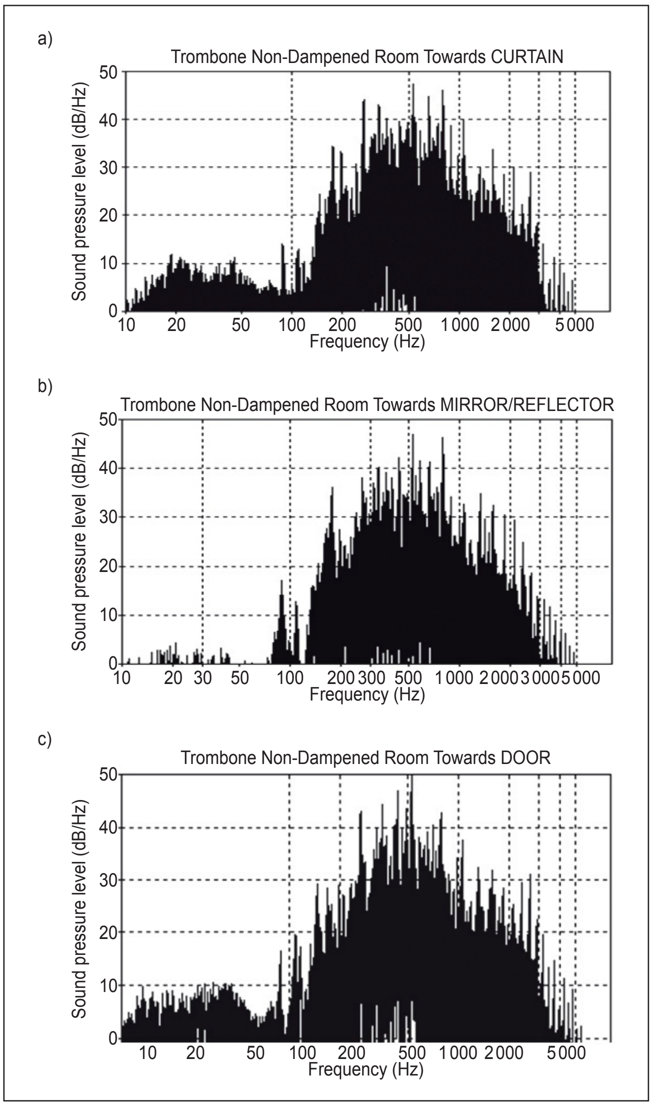


Figure 27. Frequency analysis of the composition. "In-Ear-mic". Trombone in different directions. a) Facing curtain. b) Facing reflecting wall. c) Facing door (reflecting). Logarithmic scale.

The changes in timbre between different directions are more clearly observed when we analyse the parameter Sone. Here is an analysis of the 10s forte, ***f***, section, showing Sone/Bark. Both measurements are for the "non-dampened" room. The blue curve is for the trombone facing the curtain, and the yellow is for facing the mirror.

We see an increase Sone in the Bark-region 7-12 (which might correspond to centre frequencies of 700-1600 Hz). This increase of perceived loudness is easier seen in this analysis than the traditional frequency analysis in fig. 27.

6. ROOM RESONANCES

Appendix C shows 1) Perceived room resonance in the room when singing, 2) Calculations of room resonances,

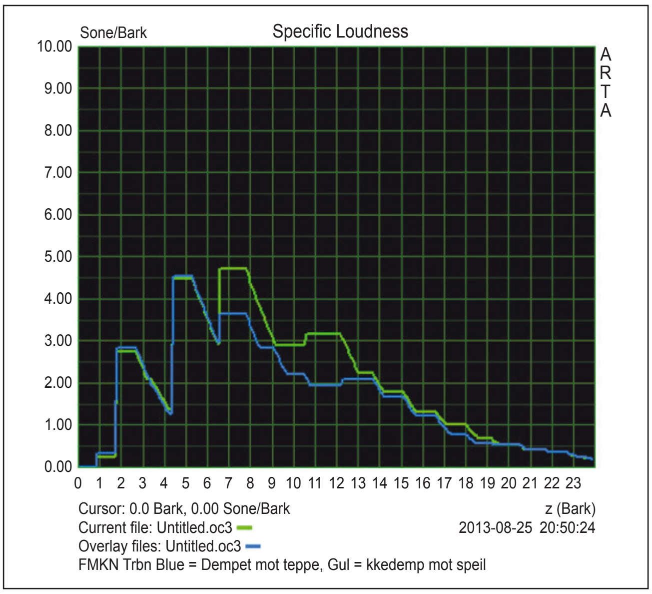


Figure 27b. Sone/Bark Trombone. "Non-dampened" room. Blue = Facing the curtain. Yellow = Facing the mirror.

3) Theoretical positions of max sound pressure levels in the room due to each resonance and 4) Practical comments on room resonances. Introducing the corner absorbent was clearly perceived as beneficial for reducing the room resonances, not only for the measured reverberation times shown earlier, but also for the overall well-being in the room, talking with a normal tenor voice (example ca. 110 Hz, see App. B). The waterfall curves without corner absorption is shown in Fig 28. (curtain only, "non-dampened").

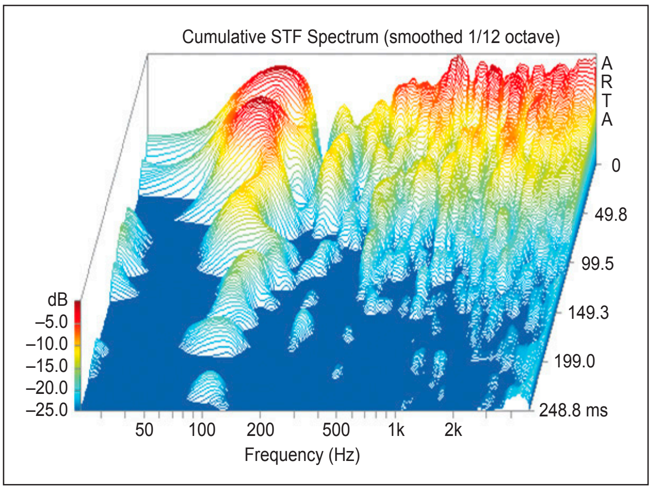


Figure 28. Waterfall Curve without corner absorption (curtain only, without corner/bass-absorber).

The corresponding Energy Decay and Schroeder-curve, filtered 1/1 octave, 125 Hz is shown in fig. 29.

The dominant "wave"-like shape of the waterfall curve (the decay around ca. 100 Hz in fig. 28) seems to "crawl" somewhat side to side, which might indicate that there are several room resonances "fighting" around the same frequency band. (see discussion in Appendix C).

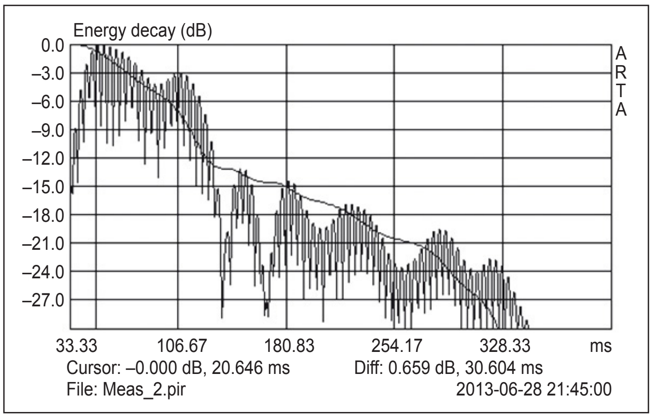


Figure 29. Energy Decay and Schroeder curve. 1/1 oct. 125Hz (curtain only, without corner/bass-absorber).

The waterfall curve with corner absorber, and wall absorber (in addition to curtain), is shown in Fig. 30.

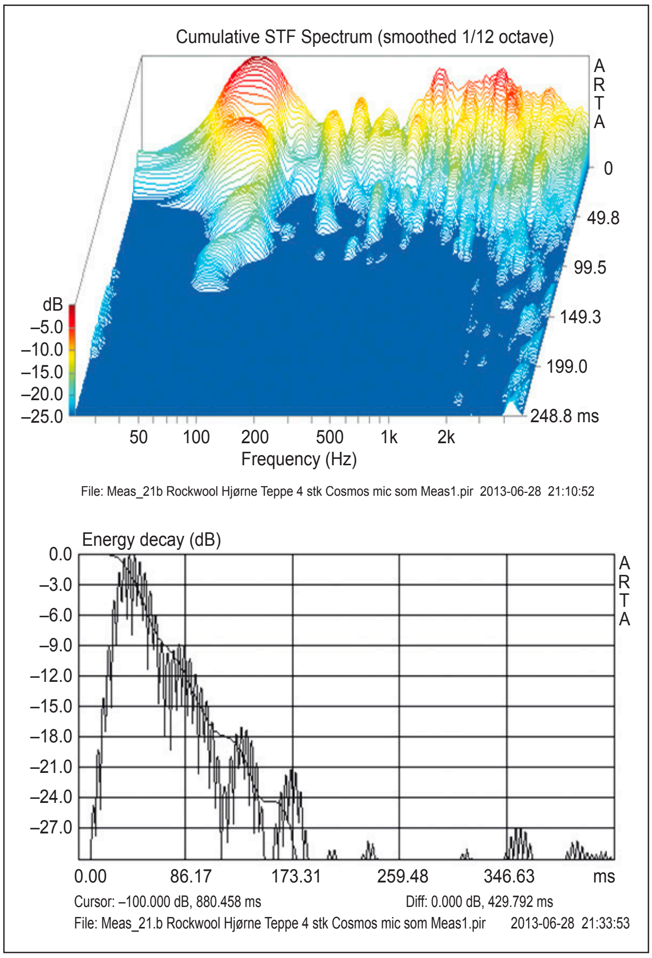


Figure 30. Waterfall and corresponding Energy Decay and Schroeder curve with corner absorber (and curtain).

We see that the corner absorber reduces the resonances. Without the corner absorption it was easy to locate perceived resonance peaks in the room "by ear" when talking/singing. This effect was reduced with the corner absorber. The tuba player was pleased with having more control in the low register. There are, however, still some resonances, mainly between ceiling and floor and between the short walls (between door and window), se App. B.

The test composition includes two sections of chromatic scales. Unfortunately, common FFT sonogram settings give inadequate resolution for such low frequencies, so there is no clear signs of specially resonating notes in the two chromatic scale passages of the Test Composition for the trombone (fig. 31 compared to fig. 32), nor any clear indication of the perceived fact that the room resonances were much less pronounced in fig. 31 with corner (and wall) absorbers.

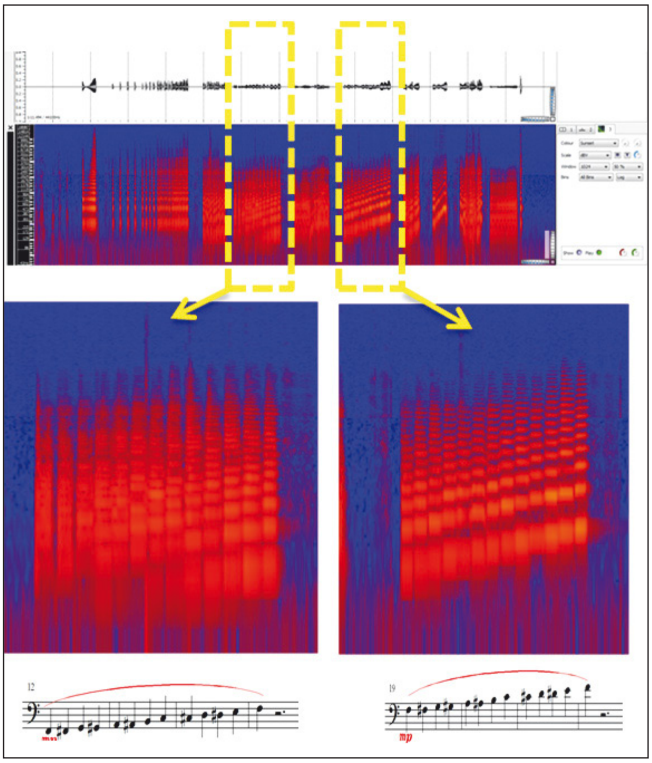


Figure 31. *Sonogram with zoom in for the two chromatic passages. Trombone. Damped room. Logarithmic frequency scale.*

Even with sharper settings for the sonogram, there are no clear signs of the influence of room resonances on strength of particular tones of the chromatic passages. (see fig. 33 and 34).

Also for tuba, the frequency resolution in this common spectrogram setting is not sufficient for checking the influence of room resonances (se fig. 35).

7. CHECK FOR COMB FILTER COLORATIONS

Coloration is clearly perceived "by ear" when singing/ talking, especially in the "non-dampened" room. As

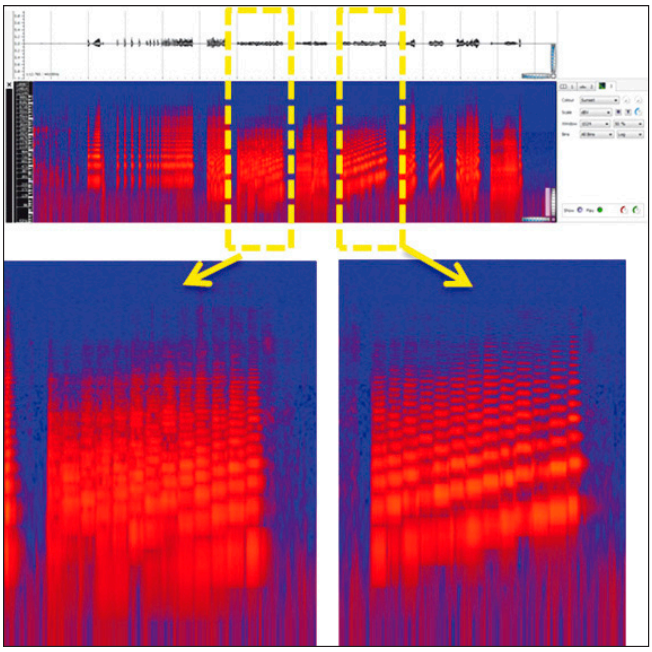


Figure 32. *Sonogram with zoom in for the two chromatic passages. Trombone. Non-Damped room. Logarithmic frequency scale.*

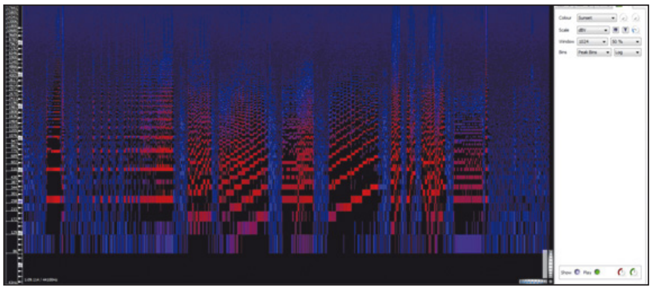


Figure 33. *Sonogram of Trombone. Sharper settings. Dampened room.*

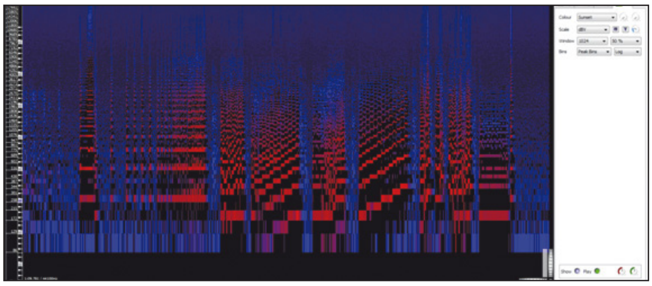


Figure 34. *Sonogram of Trombone. Sharper settings. Non-Dampened room.*

given in [3], close, discrete reflections may give comb filter coloration. This is shown for several investigations on orchestra platform in concert halls, and for echolocation for the blind [4]. Fig. 36 shows frequency analysis of "In-Ear" recordings of handclaps (linear frequency scale) in our rehearsal room. We see only small signs of comb filter coloration, even for the "non-dampened" situations. The reason might be that we have several comb filters overlapping with about the same "Comb-Between-Teeth-Bandwidths".

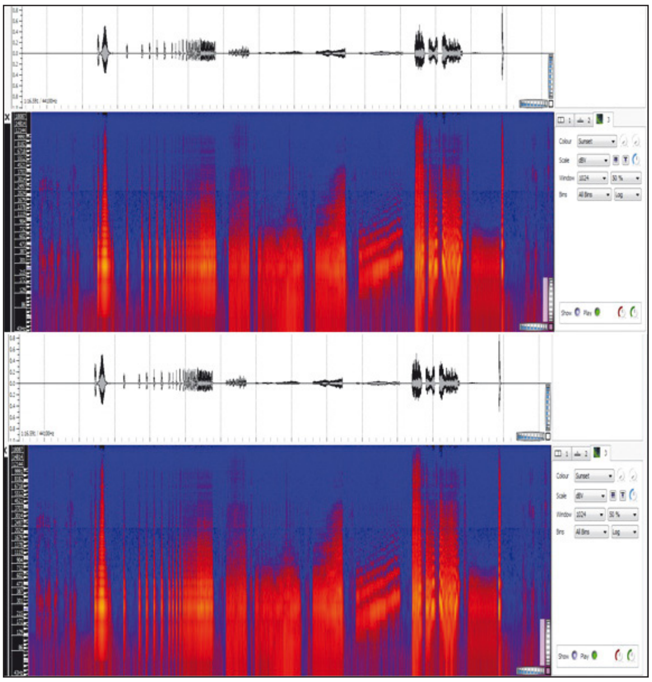


Figure 35. Sonogram of Tuba. Sharper settings. Upper: Non-Dampened room Lower: Dampened.

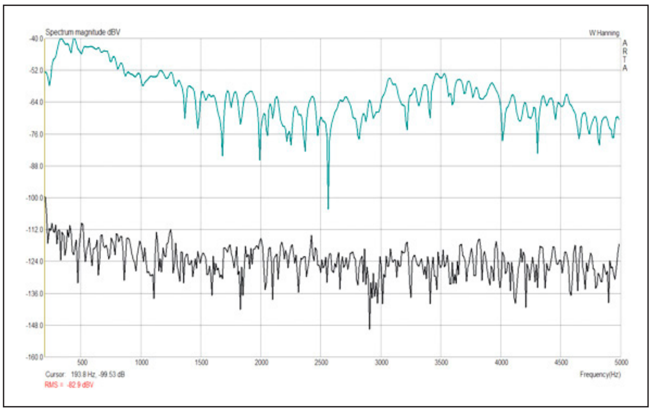


Figure 36. Frequency analysis of In-Ear recordings of handclaps. Upper: Non-Dampened room. Lower: Dampened room. Linear frequency scale.

Additional analysis showed no clear Autocorrelation for these recordings, which mean that several reflections (ceiling, floor etc.) arrive at about the same time, and that the sound field is not sufficiently correlated as to give clear comb filter effects.

8. MUSICIAN´S PREFERENCES AND PRACTICAL SOLUTIONS

For our short tests, all the musicians preferred the "dampened" settings, with all absorbers (curtain, bass-/corner-absorber and wall absorbers), but this setting might perhaps be too "dead" for long-time rehearsal. For a practical solution, the corner-/bass-absorber should be permanent (probably in the form of a slit panel in front of a corner cavity filled mineral wool), and the wall absorbers should be somewhat flexible.

The measurements and simulations were performed and analysed in: WinMLS, ARTA, Audio Tools (Studio Six) w/calibrated iMic, Sonic Visualiser, Praat, Odeon, Wavelet Sound Explorer and Pure Data (Pd)

9. CONCLUSIONS

Practical measurements and recordings of different instruments in a small rehearsal room have been performed. It is shown that a sound source of a given, constant sound power will be reduced some 3-5 dB when the room is modified from "non-dampened" (only curtain) to "dampened" (curtain, corner absorber and some wall absorbers). Musicians, however, will compensate unconsciously, so the effective, perceived reduction will in practice be some 1-2 dB lower.

The changes in timbre between a moderately dampened room and a well dampened room are not easily detected by using common settings in sonograms etc. However, for high pitched instruments, a rise in Spectral Centroid is observed due to higher relative strength of harmonics in the 2-4 kHz region which include the frequencies most dangerous for damage of hearing.

The main effect of a well absorbed room in practice is to 1) Reduce the "shimmering" of high frequencies, and 2) Dampen (some of) the room (bass) resonances, which for small rooms are in the region of the fundamentals for tenor/bass instruments.

Of course such dampening of room resonances should theoretically also give a more consistent level when playing different tones that corresponds or not to the resonances, but in practice for rehearsal, the musicians did not complain much about possible lack of egality in the frequency response, so this effect of dampening the room resonances seems not to be as important for musicians as they are for loudspeaker playback in sound control rooms of similar (small) size.

The aspects of 1) reducing "shimmering" in the high frequencies and 2) room resonances in the bass is more important than a common reverberation time approach when designing (too) small rooms for music.

10. ACKNOWLEDGMENTS

Thanks to the patient musicians and producers of the Norwegian Armed Forces´ Band North, Harstad.

11. REFERENCES

[1] Meyer, J: “Acoustics and the Performance of Music” 5th ed. Springer 2009, p.360.

[2] Halmrast, T: “When Source is also Receiver” ISRA, Int. Symp. on Room Acoustics, Toronto 2013.

[3] Halmrast, T:”*Orchestral Timbre: Comb-Filter Coloration from Reflections”.* Journal of Sound and Vibration (2000) 232(1), 53-69.

[4] Halmrast, T: *“More Combs”* Proceedings of the Institute of Acoustics (UK) Vol. 33. Pt.2, and EAA/Forum Acusticum, Aalborg, 2011.

APPENDIX A

Rehearsal Room Test
Tor Halmrast
Stright in rythm!
Flute
Clarinet in B♭
Alto Sax.
Trumpet in B♭
Trombone
Tuba
moltp, molto satcc.
marcato
sing!, dirty, growl
very smooth
GP

APPENDIX B

Room Resonances

B1) Perceived Resonance

Even in Dampened room, a (tenor) male voice gives a resonance of app. 111 Hz in the room, by singing/humming.

B2+3) Calculations and Visualisation of Room Resonances

(Black = High Sound Pressure levels)

1:1:0	90.0 Hz	corner	
0:2:1	109.2 Hz	corner	ceiling
1:0:1	110.5 Hz	corner	ceiling
1:2:0	114.2 Hz	Corner !!	
1:1:1	117.6 Hz	(corner)	(ceiling)
0:3:0	119.7 Hz	Corner !!	
etc	136 Hz		

B4) Comments on Resonances

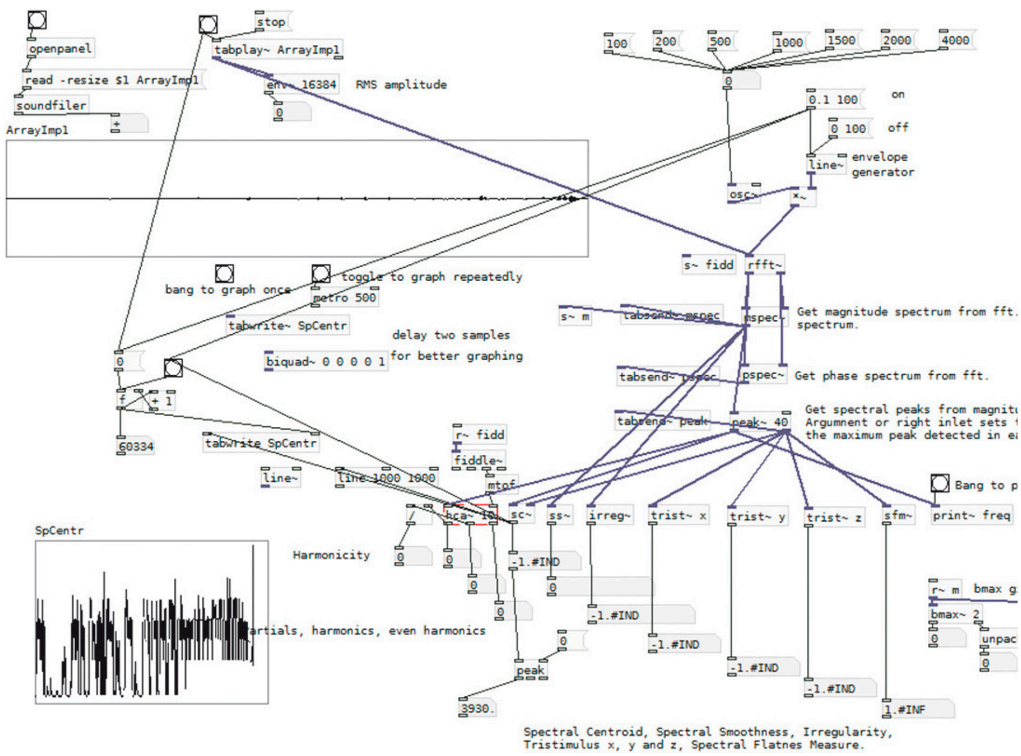
The W and H dimensions are quite close, giving almost the same frequency for two (three) resonances: 0:2:1 (109.2 Hz), 1:0:1 (110.6 Hz) and 1:2:0(144.2). The "swinging"/"crawling" shape of the resonant frequenc(ies) in the decay shown in the waterfall curves in Ch. 6 is probably because these resonances interfere.

Resonance 0:2:0 (109.2 Hz) and 1:0:1 (110.6 Hz) have almost the same as our "singing" resonance of 111Hz. Resonance 1:2:0(114.2 Hz) is the one that will be most reduced by the position of our corner absorber. The reduction of reverberation time and resonant decay in the dampened version in Ch.6 is probably due damping of resonance 1:2:0 due to corner absorber. (Our corner absorber is likely also to dampen resonance 1:0:0, (which is not so easily triggered by a (tenor) voice).

Resonances 0:2:1, 1:0:1 and specially 1:1:1 (117.6 Hz) would probably be more effectively reduced if the corner absorber could be placed all the way down to the floor. (This was not possible in our test because of the bench at the sidewall, see fig. 3)

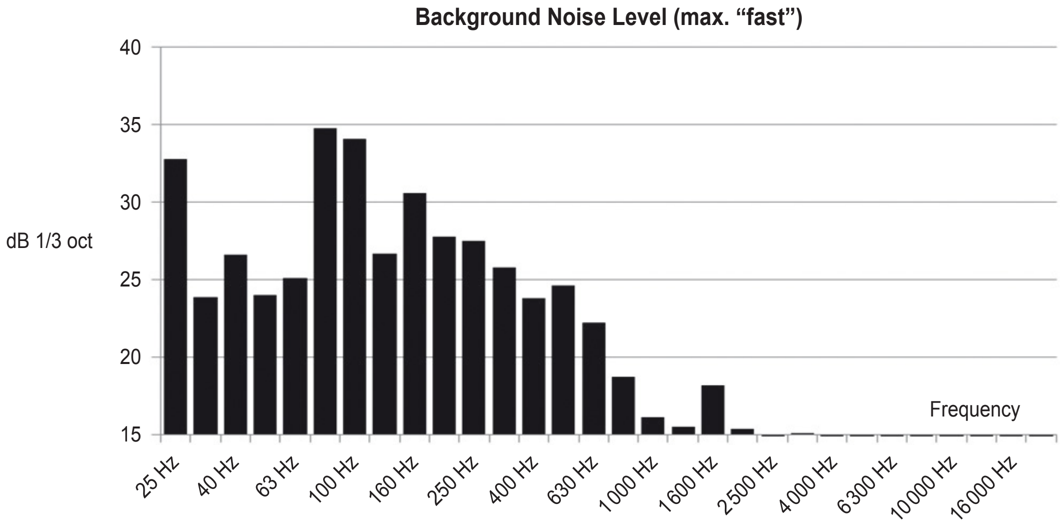
APPENDIX C

Pure Data (PD) patch for measuring Peak of Spectral Centroid



APPENDIX D

Maximum Background Noise during the measurements

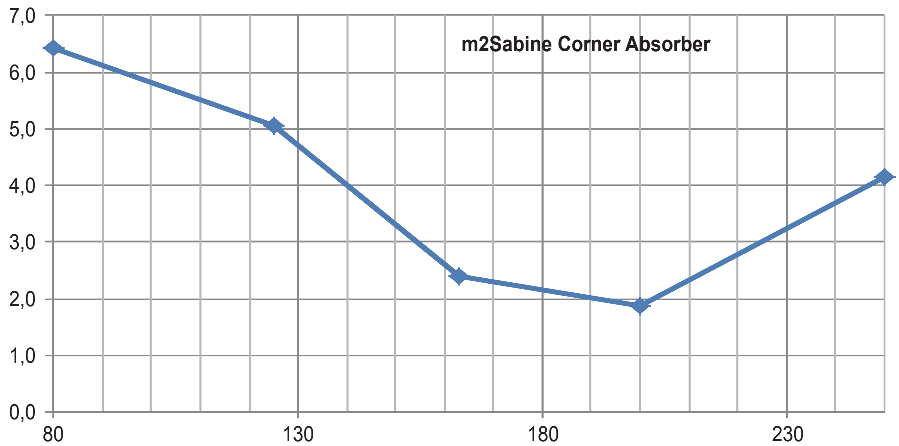


APPENDIX E

Absoption area of Corner Absorber (Roll of Rockwood)

		L	*W*	*H*	*V*	
		4,33	*2,14*	*2,28*	*21,13*	
from fig. 7	**T30**			**m2Sab**		
Hz	**No Corner Abs**	**Corner Abs**	**Diff. T30**	**No Corner Abs**	**Corner Abs**	**m2 Sab Corner Abs**
80	0,7	0,3	0,4	4,8	11,3	6,4
125	0,65	0,33	0,32	5,2	10,2	5,0
163	0,9	0,55	0,35	3,8	6,1	2,4
200	0,73	0,52	0,21	4,6	6,5	1,9
250	0,75	0,39	0,36	4,5	8,7	4,2
			mean:	4,6	8,6	4,0

T = 0,16V/A
A = 0,16V/T



The effect of scattering objects on measured reverberation times in sport halls

Don Oeters
Arup Acoustics, Winchester, UK
Corresponding author: Don.Oeters@arup.com

PACS: 43.55.-n, 43.55.Br

ABSTRACT

In a typical sports hall, the primary location for sound absorbing finishes is at high level on the walls, 'above the impact zone' and on the soffit where sound absorption is typically provided by a perforated metal deck. The measured reverberation time in sports halls can be much longer than predicted. A hypothesis is proposed that the long measured reverberation time is due to sound energy which propagates in the horizontal plane and does not 'see' the absorption at high level, therefore decaying slowly. It is proposed that diffusing or scattering surfaces can disrupt the 'horizontal sound field', re-directing sound energy to the sound absorbing surfaces. Measured reverberation times in sports halls and other rooms with and without scattering and diffusing objects demonstrate that the introduction of limited amount of scattering can result in a typical reduction in measured mid-frequency reverberation time of approximately 30%.

1. INTRODUCTION

Verification of the room acoustic design of sports halls is normally provided by the measured mid-frequency reverberation time.

Arup Acoustics data of measured reverberation times in sports halls shows that a large variation in measured reverberation time, in comparison to the predicted reverberation time is common. The large potential variation in the measured reverberation time against predicted time also ties up with the author's experience and anecdotal evidence.

This paper explores the hypothesis that the measured reverberation time in a sports hall is strongly influenced by the location of absorption in the sports hall and the scattering in the measurement plane.

2. DESIGN VERIFICATION

In the United Kingdom, the effectiveness of the sports hall acoustic design is normally assessed using the measured mid-frequency reverberation time T_{mf}. The T_{mf} is the average of the T20 or T30 reverberation times measured in the three octave bands centered on 500Hz, 1000Hz and 2000Hz. The reverberation time measurements are normally made at a height of between 1.2m and 1.5m above finished floor level.

3. HISTORICAL DATA

Arup Acoustics has accumulated a body of data of measured reverberation times. Figure 1 compares the measured T_{mf} with the calculated $T_{mf.}$

A number of calculation methods (including Sabine, Eyring, Fitzroy, CATT and Odeon) were used to generate the calculated T_{mf} for the different sports halls. This paper does not address the relationship between the calculation method and the scale of difference between the calculated T_{mf} and measured T_{mf}, although it is useful to note that where the measured reverberation time is significantly shorter for sports halls 18 and 19 the prediction method was the Fitzroy method. This paper does not address the relationship between the calculation method and the scale of difference between the calculated Tmf and measured T_{mf}. Nevertheless the body of data is considered sufficiently large that a clear relationship between measured and calculated T_{mf} times is visible.

4. CASE STUDIES

Arup Acoustics carried out a series of measurements in a sports hall and it was noted during this series of measurements that the

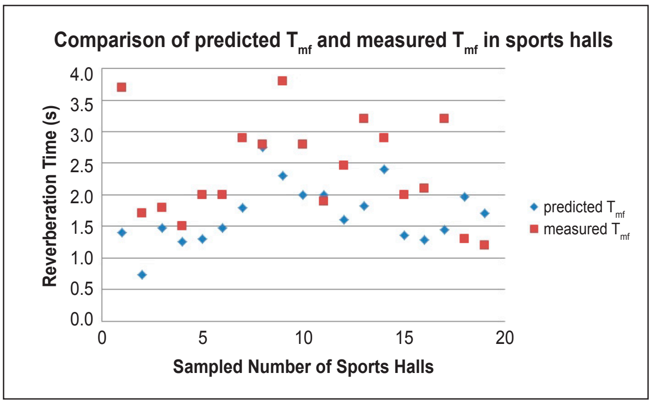


Figure 1. *Comparison of measured and predicted T_{mf}*

measured T_{mf} was significantly influenced by a number of scattering objects in the room. Figure 2 illustrates the extent of the scattering objects which resulted in a change in measured T_{mf} from 2.8s to 1.8s.

The scattering objects in the Arup sports hall study that can be seen in Figure 2 are benches and gymnasium matting stacked against the walls. The permanent absorption in the sports hall was provided by a perforated metal deck ceiling and approximately 150m^2 of wall mounted absorption.

Following these results, Rob Cornetta of London Southbank University carried out a measurements of reverberation time in a sports hall. A change in measured T_{mf} from 4.0s to 3.1s was recorded when scattering objects were introduced into the room. Further information regarding these measurements is available at www.lsbu.ac.uk/isess/.

The scattering objects in the London Southbank University study are shown in the lower row of photographs in Figure 3 and consist of table tennis tables and gym mats stacked against the walls.

Figure 3. *Southbank University study, extent of scattering objects in sports hall resulting in change of measured T_{mf} from 4.0s to 3.1s.*

5. OTHER REPORTED MEASUREMENTS

A study of school acoustic conditions was reported in IOA Bulletin March / April 2010[1]. During this study the T_{mf} was measured in a sports hall. The first set of measurements was carried out with desks and chairs for examinations in the hall before and after installation of sound absorbing treatment. The T_{mf} was measured again after the removal of the furniture and it was reported that:

'*With plastic and metal chairs in the room the reverberation time was reduced from 2.0s to 1.2s, but with no furniture the reverberation time post- treatment was 2.8s, significantly worse than before treatment'.*

Erling Nilsson, of Ecophon, carried out research into the effect of scattering objects, which is reported in NT Technical Report TR606, 2007, Sound Scattering in Rooms with Acoustic Ceiling Treatment [2].

The research investigated the effect of introducing tables and chairs arranged as classroom furniture into

Figure 2. *Arup case study, extent of scattering objects in sports hall resulting in change of measured T_{mf} from 2.8s to 1.8s.*

an otherwise plain room with a sound absorbing ceiling.

This research showed a reduction in the order of 30% to 40% in measured T_{mf} upon the introduction of furniture into the test room.

6. SUMMARY OF THE EFFECT OF SCATTERING OBJECTS IN THE ROOM

The measured effect of scattering objects can be summarised as follows;

The measured T_{mf} in rooms with absorption only on the ceiling or otherwise away from the measurement plane was between 25% and 50% shorter than when furniture / scattering was introduced into an otherwise plain room.

Or:

The measured T_{mf} in rooms with absorption only on the ceiling or otherwise away from the measurement plane was between 30% and 100% longer when furniture / scattering was removed leaving a plain room.

7. HYPOTHESIS BASED ON GATHERED DATA

Based on the evidence gathered of measured reverberation times, it is proposed that that:

‘When absorption is only located on the ceiling or otherwise away from the measurement plane, and there is insufficient acoustic diffusion / scattering in the horizontal plane, sound energy propagating in the horizontal plane is not incident on the absorption which can result in a ‘long’ measured reverberation time. ‘

This conclusion infers that when the sound absorption is located only on the ceiling or otherwise away from the measurement plane, then for the absorption to be effective, scattering objects are needed in the room to redirect sound energy so that it is incident on the absorption in the room.

8. IMPLICATIONS

The conclusion, that the measured reverberation time can be so heavily influenced by scattering objects in a room, has a significant impact on the notional accuracy of calculations of reverberation time, since:

- Sabine/Eyring prediction method-relies on diffuse sound field
- Fitzroy method-relies on absorption in source plane
- ray-tracing-relies on accurate assignment of diffusion to wall surfaces

Since in a plain room with absorption located on the ceiling:

- the sound field may not be diffuse
- measurement of reverberation time is made in a horizontal plane which contains little acoustic absorption
- the quantitative diffusion of the walls in the horizontal plane of measurement may not be known,

It is implied that the measured T_{mf} may typically be up to twice as long as the calculated T_{mf}. This gives a large uncertainty to the calculated T_{mf}.

9. CONCLUSION

The measured T_{mf} is currently used to assess that sufficient sound absorption has been included in rooms such as sports halls.

In orthogonal rooms, with absorption on the soffit, the measured T_{mf} is strongly dependent on the scattering in the horizontal measurement plane.

There is no reliable method for assigning scattering or diffusion to objects or surfaces in the room and the absorption is located away from the measurement plane.

In tall spaces with few or no scattering objects, the measured T_{mf} is not accurately predictable and therefore may be considered unsuitable for verification of the acoustic design.

REFERENCES

[1] IOA Acoustic Bulletin Vol35 No2 March / April 2010 pp34-38.

[2] E. Nilsson: NT Technical Report TR606, 2007, Sound Scattering in Rooms with Acoustic Ceiling Treatment.

Extracting meaningful uncertainty data from Calibration Certificates and associated Sound Level Meter Standards

Ian Campbell
Campbell Associates, UK.
Corresponding author: ian@campbell-associates.co.uk

PACS code: 43.38.Kb, 43.58.Vb

ABSTRACT

Any measurement made for regulatory or other legal reasons now comes into the general field of Legal Metrology and the uncertainty associated with the measurement has to be evaluated. This is particularly true if the results of the measurement exercise are to be challenged in either an enquiry or court proceedings. All "noise" measurements have a number of key elements that need to be considered when considering the uncertainty that will be associated with the result. These include the variability of the source, effects of the transmission path, the environment and of course the measurement instrument its self. This paper considers the impact of the instrument, its standardisation and maintenance and how these elements can be quantified.

The measurement instrumentation is a natural element within the total uncertainty budget associated with a "noise assessment". With correct standardisation, maintenance and field calibration routines it is one that can readily be controlled and quantified. As always the devil is in the detail as sound level meters are required to quantify the sound whist most audio kit is just required to reproduce it, either for communications or entertainment purposes. For these latter applications the end result is judged subjectively and so changes of a few dB are not going to be noticed. With sound level meters these few dBs could be very significant in the overall project that the measurement is designed to support. For general audio applications wide tolerances on system stability, frequency response and dynamic range are acceptable; however for quantifying the sound field these parameters have to be closely controlled over a wide range of environmental conditions.

Historically there has been an attitude of Caveat Emptor (Buyer is responsible for the validity of the manufacturer's claims) and this has allowed a large number of sound level meters to be introduced without any verification that they actually do what is expected of them. Over the past fifteen to twenty years there has been a development of the established "weights and measures" inspection procedures into a system of Legal Metrology that will deliver a high degree of confidence in the performance of measurement systems. These comprise of basic standards (BS EN ISO etc.) that determine what the instrument should do; these are supported by three levels of confirmation.

1. Firstly there are the "Pattern Evaluation" requirements that provide for an independent testing regime that will confirm that the basic design of the instrument does what the specification requires over the complete range of environments and measurement conditions covered by the basic specification. The manufacturer normally sends 3 to 5 samples of the instrument to a National Testing Laboratory (NPL, PTB, LNE etc.) who will issue the necessary confirmation that the basic design of the instrument allows it to meet its claims.

2. Secondly there is the "Periodic Verification" which is an annual or bi-annual examination by an accredited laboratory where a sub-set of the pattern evaluation tests are performed to confirm that the instrument is still within its original calibration limits.

    a. If the pattern evaluation has been carried out by a National Metrology Laboratory and the periodic verification by an

Accredited Laboratory then a certificate of calibration and conformance to the standards can be issued as opposed to just a certificate of calibration

3. Finally the user of the instruments will find details in the instruction manuals of the field checks that they should carry out before and after each set of measurements to confirm that the kit is working correctly. These include use of a sound calibrator, checks on the power supplies, range settings, self-noise, etc.

In respect of sound level meters and sound calibrators the governing standards for these various levels of verification are set out below.

Pattern evaluation is the responsibility of the manufacturer in conjunction with a National Metrology Laboratory and the Periodic Verification is the responsibility of the user in conjunction with an accredited calibration laboratory.

Earlier versions of the standards had comments about the basic accuracy of the sound level meter of ±0.7 dB for a class 1 device and ±1 dB for a class 2. This however related to the accuracy at reference level and reference frequency (normally 114 dB at 1k Hz) as well as at reference conditions (23°C, 101.325 kPa and 50% RH). As these conditions hardly ever exist in relation to a practical noise measurement they are not very helpful and so they do not appear in the more recent version of the standards.

There are two elements to the measurement kit to be considered both independently and how they work together. These are the sound calibrator and the sound level meter.

The periodic verification requirements for a sound calibrator are for there to be 3 replications, to give a measure of how good the fit to the microphone, a measurement of the short term stability of the output, a measurement of the frequency and the frequency stability as well as a measurement of the total distortion. An example of these results is given below.

The standard deviation of the three replications is combined with the uncertainty of the basic measurement (Expanded Uncertainty row) to give the degree of freedom and coverage factor. If this figure is 2 then there is 95% confidence that the result lies within the basic uncertainty figure. If there was a poor fit to the microphone then the three independent measurements would differ and the coverage factor would move away from the required result and it would not be possible to use the basic accuracy of 0.1 dB in the overall measurement uncertainty calibration and a higher figure needs to be calculated.

The figure arrived at would be safe to use if the sound level meter and calibrator were paired during the periodic verification and environmental conditions were close to reference. If however a different calibrator is being used then additional corrections would be needed for the pressure to free field correction, volume loading effect and environmental characteristics.

The pressure to free field corrections and volume loading are often combined and quoted in calibrator manuals as the correction to be applied for that specific combination of microphone and preamplifier, typically 0.2 dB at 1k Hz. Environmental correction will be a combination of the effects on the sound calibrator and the microphone whilst for most meters the effect of the electronics and humidity can be discounted. The table below gives details for the combination of a typical class 1 meter and calibrator combination and based on information like this a combined environmental uncertainty of around 0.3 dB over the specified range of environments would be reasonable. If however the information is not available then the calculation would have to be based on the limit values given in the specification which would double this figure.

The Nor-1251 is a “smart calibrator” having internal compensation for environmental and volume loading effects; should an uncompensated calibrator be used (designations L or C under the standard) then corrections should be applied to bring performance into line. The uncertainties of these corrections are not quantified at the moment.

Instrument Type	Valid dates	Specification	Pattern Evaluation	Periodic Verification
Sound Level Meters	1979 to date	BS EN 60651 & 60804	OIML[1] R58 & R88	BS 7580
	2003 to date	BS EN 61672-1	BS EN 61672-2	BS EN 61672-3
	2013 to date	BS EN 61672 v2 -1	BS EN 61672 v2 -2	BS EN 61672 v2 -3
Sound Calibrators	2003 to date	BS EN 60942	BS EN 60942 Annexe A	BS EN 60942 Annexe B

Figure 1. *Sound Level Meter Standards.*

[1] Organisation International de Metrologie Lègal.

Measurement Results:	Level	Level Stability	Frequency	Frequency Stability	Distortion
1:	114.00 dB	0.01 dB	1000.74 Hz	0.00 %	0.35 %
2:	114.00 dB	0.01 dB	1000.73 Hz	0.00 %	0.34 %
3:	113.99 dB	0.01 dB	1000.72 Hz	0.00 %	0.34 %
Result (Average):	**114.00 dB**	**0.01 dB**	**1000.73 Hz**	**0.00 %**	**0.34 %**
Expanded Uncertainty:	0.10 dB	0.02 dB	1.00 Hz	0.01 %	0.10 %
Degree of Freedom:	>100	>100	>100	>100	>100
Coverage Factor:	2.00	2.00	2.00	2.00	2.00

Figure 2. *Typical statement of results for a sound calibrator as per Annex B of BS EN IEC 90942.*

Model number	Temperature, °C			Barometric Pressure, kPa			
	Coefficient	Ref =	23	Coefficient		Ref =	101.325
		–10	50		65	85	108
Nor-1225 Microphone	–0.007	0.23	–0.19	–0.01	0.36	–0.16	–0.07
Nor-1251 Calibrator	0.003	–0.10	0.08	0.005	–0.18	–0.08	0.03
Result		**0.13**	**–0.11**		**0.18**	**–0.24**	**–0.03**
Acceptance interval specified		**0.5**	**–0.5**		**0.9**	**0.4**	**0.4**

Figure 3. *Typical environmental effects on the basic calibration setting of a sound level meter.*

Model number	Temperature, °C			Barometric Pressure, kPa			
	Coefficient	Ref =	23	Coefficient		Ref =	101.325
		–10	50		65	85	108
Nor-1225 Microphone	–0.007	0.23	–0.19	–0.01	0.36	–0.16	–0.07
Gras 42AA Pistonphone	0.0005	–0.02	0.01	0.0853	**–3.10**	**–1.39**	**0.57**
Result		**0.21**	**–0.18**		**–2.74**	**–1.56**	**0.50**
Acceptance interval specified		**0.5**	**–0.5**		**0.9**	**0.4**	**0.4**

Figure 4. *Environmental effects when an uncompensated calibrator is used.*

Having determined the accuracy of the setting of the meter at the reference level and frequency it is then necessary to consider the performance at other levels and frequencies.

The system frequency response is predominantly determined by the microphone and the effects of the case etc. The example below is for a class one device shown with its tolerances over the normal test range of 100 to 20k Hz. The three stars show the only points at which the standard mandates an acoustic measurement of the frequency response; the rest of the data is optional, including the extension down to 2 Hz. These results confirm that over the central range of 25 to 8k Hz this individual device is essentially flat and additional uncertainty elements are only needed if there is significant content outside this frequency range.

The natural frequency response of the system will be post processed to provide the frequency weightings A, C and Z. These will be tested electrically and then these results are combined with the acoustic tests made on the microphone along with the correction data provided by the manufacturer for the case reflection and the effects of any accessories such as a windscreen. The acceptance interval quoted in the standards applies to the overall instrument and accessories and not just the microphone. These combined results should be inspected to confirm that there are no significant deviations from the required response. In the example shown as figure 6, which

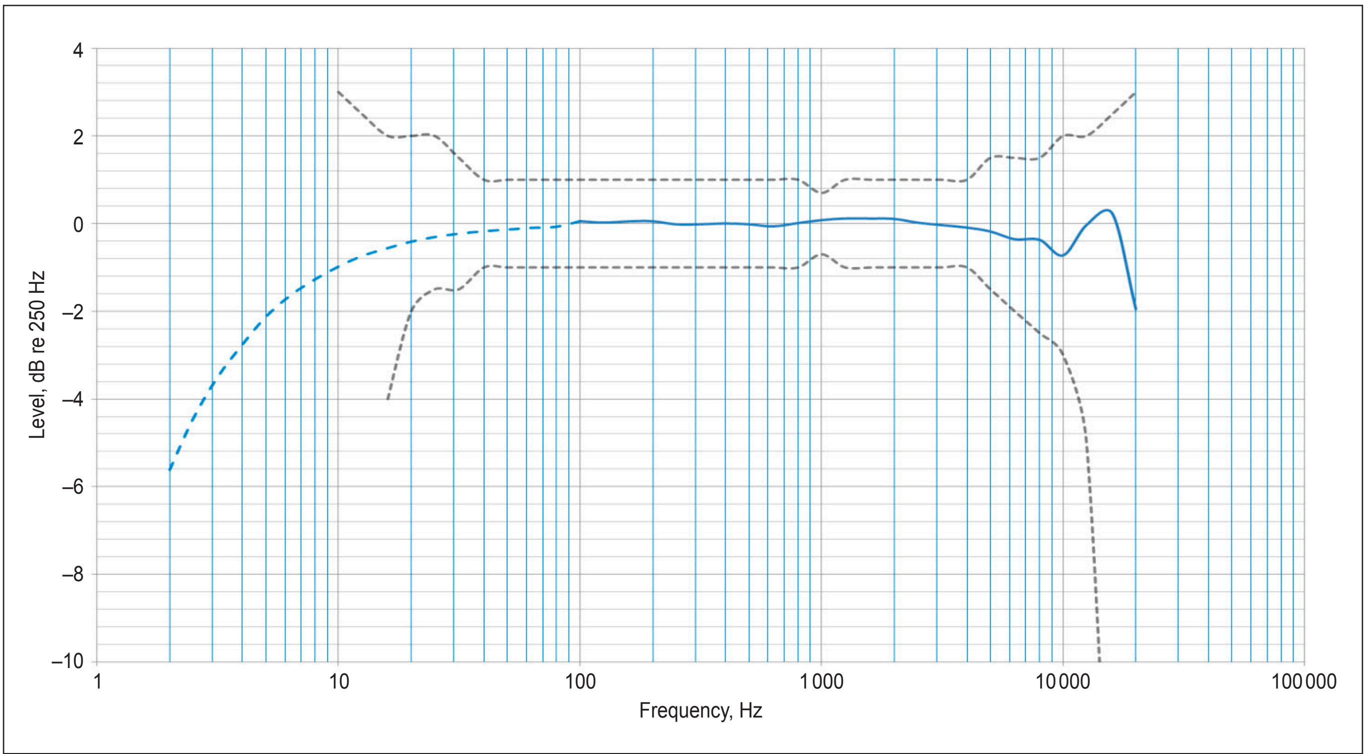


Figure 5. *Typical microphone response, with optional LF extension.*

shows the combination of the electrical, acoustic and accessory data, it can be seen that the results for frequencies above 4k Hz would need to be corrected in the results or additional elements added to the uncertainty computation to take account of the errors in these frequency bands. The optional extension of the results to 2 Hz also show deviations and these too would have to be treated in a similar manner.

As amplitude linearity is normally very good in modern sound level meters it is only really necessary to consider the limit values, the minimum values will be controlled by the self-noise and the maximum by the overload point.

Most modern sound level meters have a single range covering over 120 dB and have their overload point set to 130 or 140 dB so this does not usually cause a problem. Should the meter show an overload then its effect on the measurement needs to be considered. It may not be significant if the object was to determine an L_n background level but is very significant in a L_{Aeq} or L_{Cpk} measurement; correction data is not usually available in these latter cases so they need to be avoided. At the bottom end there are two elements to the self-noise; firstly there is the electrical noise of the preamplifier and secondly the microphone capsule its self. The electrical noise is measured during the periodic verification using a dummy microphone and is usually around 10 dB(A) however most half inch 50mV/Pa free field measurement microphone capsules have a self-noise of around 14 dB(A) and these two figures need to be combined to give the limiting level to which measurement can be made. With a 10 dB margin added to the combined level this gives a minimum level of 25 dB below which additional uncertainty elements need to be considered. Within this total amplitude span for the measurement there are allowable limits for amplitude linearity that would need to be considered.

The final consideration is the computation of the actual measurement metrics; L_{AF}, L_{AS}, L_{Amax}, L_{Cpk}, L_{Aeq} etc. A series of tests have been devised to check these parameters by means of gated electrical signals having known relationship to the noise indices to be verified. The first step is normally to determine the accuracy of the time constants Fast and Slow. This is achieved by examining their response to signal bursts. The rise time is determined by the time constant specified for each test and hence a single test gives a good guide to the accuracy of the RMS calculation. In the example shown the results for these two basic time constants are in order. Note that for impulse time constant the test has to be more comprehensive as in this case the rise and fall times are different.

The Peak time constant is the one that normally needs special attention as it is the metric that is specifically mandated in the Noise at Work Regulations; it is an offence to expose any employee to levels above 137 L_{Cpk}. Basically it is tested by comparing the response to a very short burst to a longer one. The result shown is

Combined acoustic and electrical results

A-Weighted results

Freq. (Hz)	SLM (dB)	Mic. (dB)	Case Ref. (dB)	Wind Scr. (dB)	Tol. (dB)	Dev. (dB)
31.5	–0.1	0	0	0	–1.5	–0.1
63	–0.1	0	0	0.1	–1.5	0
125	0	0	0	0	–1	0
250	–0.1	0	0	0	–1	–0.1
500	0	0	0.1	0	–1	0.1
1000	0	0	–0.1	0.1	–1	0
2000	–0.1	–0.3	0.2	0.3	–1	0.1
4000	–0.1	–0.9	0	0.7	–1	–0.3
8000	0	–1.2	0	–0.1	+1.5,–3	–1.3
12500	0	–0.8	0.1	–0.5	+3,–6	–1.2

C-Weighted results

Freq. (Hz)	SLM (dB)	Mic. (dB)	Case Ref. (dB)	Wind Scr. (dB)	Tol. (dB)	Dev. (dB)
31.5	–0.1	0	0	0	–1.5	–0.1
63	0	0	0	0.1	–1.5	0.1
125	0	0	0	0	–1	0
250	0	0	0	0	–1	0
500	0.1	0	0.1	0	–1	0.2
1000	0	0	–0.1	0.1	–1	0
2000	0	–0.3	0.2	0.3	–1	0.2
4000	–0.1	–0.9	0	0.7	–1	–0.3
8000	0	–1.2	0	–0.1	+1.5,–3	–1.3
12500	0	–0.8	0.1	–0.5	+3,–6	–1.2

Z-Weighted results

Freq. (Hz)	SLM (dB)	Mic. (dB)	Case Ref. (dB)	Wind Scr. (dB)	Tol. (dB)	Dev. (dB)
31.5	–0.1	0	0	0	–1.5	–0.1
63	–0.1	0	0	0.1	–1.5	0
125	0	0	0	0	–1	0
250	0	0	0	0	–1	0
500	0	0	0.1	0	–1	0.1
1000	0	0	–0.1	0.1	–1	0
2000	0	–0.3	0.2	0.3	–1	0.2
4000	0	–0.9	0	0.7	–1	–0.2
8000	0	–1.2	0	–0.1	+1.5,–3	–1.3
12500	0	–0.8	0.1	–0.5	+3,–6	–1.2

Test Passed

Colour	Meaning
Yellow	Results obtained by electrical testing
Green	Results of acoustic testing, some labs use nominal data for these tests
Blue	Correction data provided by the manufacturer, without uncertainties!

Figure 6. *Combining Electrical, Acoustic and Manufacturer's data to determine complete instrument response.*

Time weighting F & S

Time constant (ms)	Burst duration (dB)	Reference value (dB)	Measured value (dB)	Tolerance value (dB)	Error value (dB)	Result
Fast	200	112	111.9	1	–0.1	P
Slow	500	108.9	108.9	1	0	P

Figure 7. *Tests for time weightings.*

Peak response							
Pulse duration	Pulse polarity	Reference value	Measured value	Tolerance value	Error absolute	Error relative	Result
(ms)	(dB)	(dB)	(dB)	(dB)	(dB)	(dB)	(dB)
10	+	116	116.2	0.2			
0.1	+	116	115.1	2	–0.9	–1.1	P
10	–	116	116.2	0.2			
0.1	–	116	115.1	2	–0.9	–1.1	P

Figure 8. *Peak response test results.*

Time averaging, L_{Aeq}					
Burst value	Reference value	Tolerance value	Measured value	Error value	Result
(ms)	(dB)	(dB)	(dB)	(dB)	
1/10^3	107	1	106.9	–0.1	P
1/10^4	97	1	96.8	–0.2	P

Figure 9. *Time averaging test results.*

from the BS EN IEC 60651 standard and shows that this particular meter is reading 1 dB low in the peak mode hence this should be accounted for when risk assessments are being made using this particular instrument.

Following on from the calculation of the time domain indices are computed and the most common one is the $L_{eq,t}$. This result is very much affected by the peak values logged during the measurement so it seems reasonable to test using short duration pulses. The basic test is that the pulse mark space ratio is increased by a factor of 10 and this will result in a reduction in the $L_{eq,t}$ value of 10 dB.

The other time history metric in wide use is the Ln value; at the current time these are not covered by the sound level meter standards but there are a few regulations that specify how they should be calculated, i.e. the sample rate, time constant used and bin size for the classification of the distributions. There is a DIN standard that specifies these tests and this is used for verfication when clients require it to be done.

It is good practice to keep a running log of the calibrator's periodic verification as this provides evidence on which to base the recommended recalibration interval. An example is given in Figure 10 below showing the value returned at each calibration over a 16 year period. The error bars on each measurement show the uncertainty of the laboratory performing the verification. With data like this it is possible to use an uncertainty of 0.1 dB for the basic sensitivity setting of the meter and calibrator combination; we have seen examples where this could

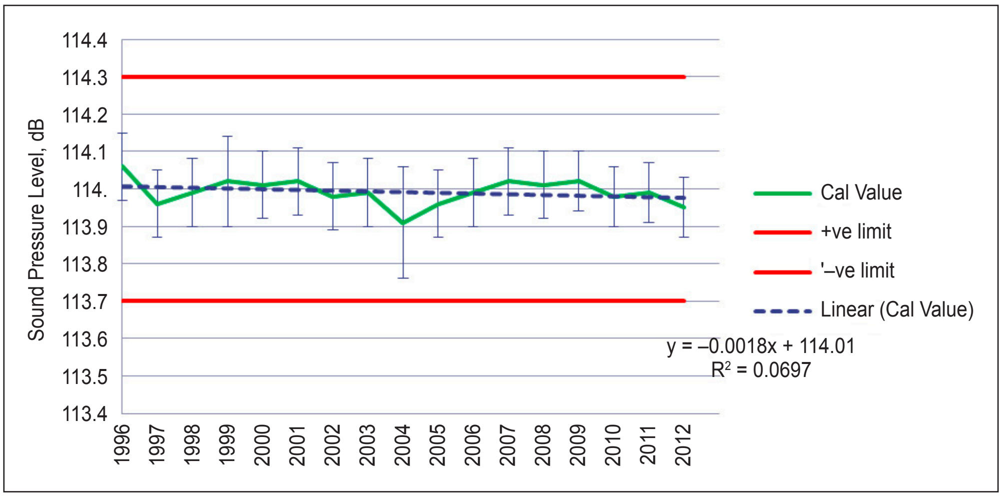


Figure 10. *Calibrator periodic verification record for drift analysis.*

extend to 0.25 dB or higher where the laboratory uncertainty or the fit of the devices is not as good as this example.

REFERENCES

– BS EN IEC 60651:1994 Specification for Sound Level Meters.

– BS 5969:1991 Specification for Sound Level Meters.

– BS EN IEC 60804:2001 Integrating-Averaging Sound Level Meters.

– BS EN IEC 60942:2003 Sound Calibrators (Specification, pattern evaluation & periodic verification).

– BS EN IEC 61672-1:2003 Sound Level Meters, Specifications.

– BS EN IEC 61672-2:2003 Sound Level Meters, Pattern Evaluation.

– BS EN IEC 61672-3:2006 Sound Level Meters, Periodic Tests.

– BS 7580:1997 Specification for the verification of sound level meters.

– OIML R58 1998 Sound Level Meters, Legal Metrology Requirements.

– OIML R88 1998 Integrating Averaging Sound Level Meter, Legal Metrology Requirements.

An eye-witness report on how the CD came about

Jacques Heemskerk
Former Philips employee
Corresponding author: jpj@famheemskerk.nl

PACS: 43.38.-p.
Originally published at European Physical Society, EPN 44/6, p.21, 2013

ABSTRACT

The Compact Disc was a smashing success. It pushed the traditional 45 and 33 RPM vinyl records off the market in an astonishingly short period of time. But the struggle to develop the CD and to get it on the market was just as fascinating.

1. INTRODUCTION

The various technologies employed in the Compact Disc (CD) player, including laser optics, a-spherical lenses, digital signal processing, integrated circuitry, nano-scale injection molding are truly revolutionary by itself. Interestingly, the basic idea to write and read information to and from a disc by optical means was not new, as can be seen in Fig. 1 taken from a 1931 article published in *Funkschau*. The pictures are not very clear, as would have been the signal produced for being limited by noise, preventing practical implementation of this reflective disc. Nevertheless, the transmission version of this scheme has been successfully employed in reading the sound track of the first "speaking pictures".

The optical scheme of the Compact Disc player is very similar to the one published in *Funkschau*, see Figure 2. However, the devil is in the detail. In a CD player, the light source is a laser, *i.e.,* a real point source. So, in case of perfect optics, the read-out spot, which is the image of the point source on the disc, is of minimum, diffraction limited size. In formula, $D = \lambda/NA$, with D the spot diameter, λ the laser wave length, and NA the numerical aperture of the objective lens (NA equals the refractive index n times the sine of the convergence angle α).

The Compact Disc itself looks quite different from the 1931 example. The audio signal is digitally coded instead of stored as the audio wave itself. The signal is not derived from reflectivity variations in a photographic material but from a relief pattern, a succession of "pits"

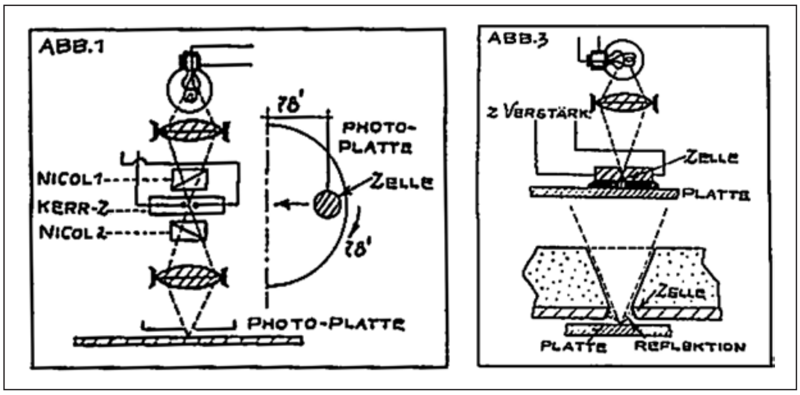


Figure 1. *Original 1931 figures, showing the idea behind the reflective audio disc recorder.* ***Left:*** *Recording method. Lamp light is focused onto a small spot on the disc where the spot size is determined by the pinhole just above the information plane. A Kerr cell is placed in the optical path between two crossed polarizers. By applying a voltage across the Kerr cell generated by the audio signal, the intensity of the light spot on the disc is modulated and the audio signal is recorded onto the photographic material.* ***Right:*** *Reading set up. The lower part of the panel shows the details of the detector geometry. The spot size is limited by a pinhole. Variations in local reflectivity are measured by a ring-shaped detector, in this case a photoelectric Selenium detector.*

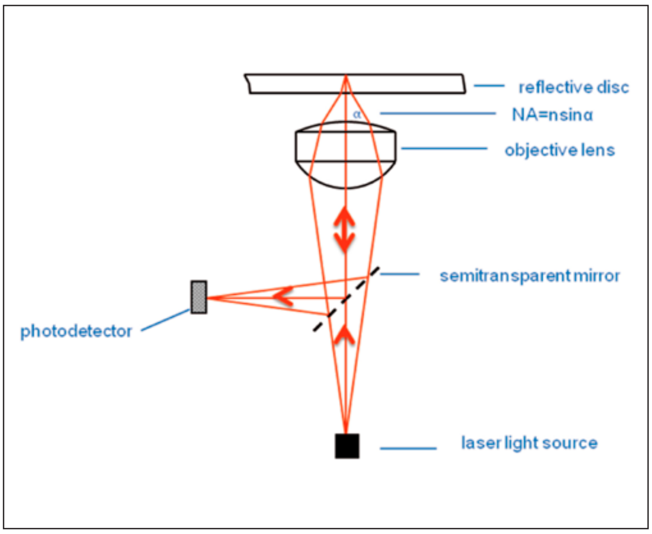


Figure 2. *Optical scheme of the Compact Disc player.*

and “lands”. Digital coding means that the audio signal has undergone a series of processing steps: first, the audio signal is transformed into 16-bit samples; second, parity bits are added (for error detection and correction); and third, the bit train is transformed into an information track on the disc, consisting of pits and lands of discrete lengths of 3, 4, … ,11 units, where one unit corresponds to 0.3 μm on the disc. The bit rate and bit density are constant. This means that the scanning speed is constant (1.3 m/s), and that the angular frequency of the disc gradually decreases during play-back (the reading head is moving from inside to outside of the disc).

2. TWO COMPANIES, TWO CULTURES

In this brief history, I should like to share my personal experience of CD development. Such experience often makes innovation real fun. I myself became involved in CD development in 1979, directly after Philips gave its first public demonstration of a CD prototype player and after Philips had toured Japan in search of allies to join into the development of the CD as a successor of the LP record. Sony was the company that responded most eagerly to this invitation, and Philips and Sony agreed on a joint development of a miniature-sized digital audio disc. The two companies realized that a common format would be necessary to convince the other Consumer Electronics companies to join at a later stage. As an optical engineer, I was present at all meetings between Sony and Philips to forge a common format.

I remember our first technical meeting very well. Management had instructed us to be completely open: “one cannot blow with one’s mouth shut”. However, we as scientists and engineers, knew very well how much effort it had cost us to achieve a reliable prototype and —as is often the case amongst engineers— we doubted whether our management really knew what they were doing. The more so, because Japan at that time was in a similar position as China is today: its industry was growing fast and it was changing from a low-cost producer into a high-tech competitor. So, at that first meeting, mistrust was the overriding emotion at our side. Later, I heard from my Japanese counterparts that they had similar reservations about the wisdom of their own management.

Looking back, I think our first meeting actually went pretty well, although we did not achieve very much in technical terms: we made a good start in team building with engineers that belonged not only to a different company, but also to a different culture. We succeeded in doing so at a time (1979) when the world was certainly not “globalized”. Initially, I felt as if those guys came from another planet. Indeed, they came from “the Far East”. Soon however, we discovered that engineers are pretty much the same all over the world, with similar emotions: proud of their expertise, sincere about their problems, eager to explain their own solutions, and willing to learn and to appreciate the solutions of others. Soon, we found a way of working together that turned out to be very effective: we challenged each other’s assumptions, and we agreed that we should not accept technical proposals out of politeness or kindness: we accepted only the best solutions, based on data, not on theories.

We quickly established a pattern of regular 2- or 3-monthly meetings, either in Japan or in Holland, and within 2 years we reached a complete agreement on the format that is now called the CD. The problem we had to solve was rather straightforward: find the best solution in reliability and information density for a small digital, read-only disc. The technical solution was less straightforward: while we both had developed our prototypes step-by-step, when reconsidering our solutions we realized that every new choice was leading to another new issue. Obviously, the most direct solution to increase information density is to increase the NA, and to reduce the wave length of the optics. However, as Fig. 3 shows, the disc substrate is part of the objective lens, so any variation in thickness will cause spherical aberration ($\Delta d \times NA^4/\lambda$), and any tilt angle will cause coma ($\varepsilon \times d \times NA^3/\lambda$) with Δd the thickness deviation, and with ε the tilt angle of the disc substrate. Because of the strong dependence on NA, a small increase in NA directly results in more stringent disc tolerances. As a result, we had very tough discussions about realistic disc tolerances, because we had only a few preliminary disc samples available of rather uncertain quality. In the end, we agreed on NA= 0.45, but with strong reservations on both sides, especially from our media engineers.

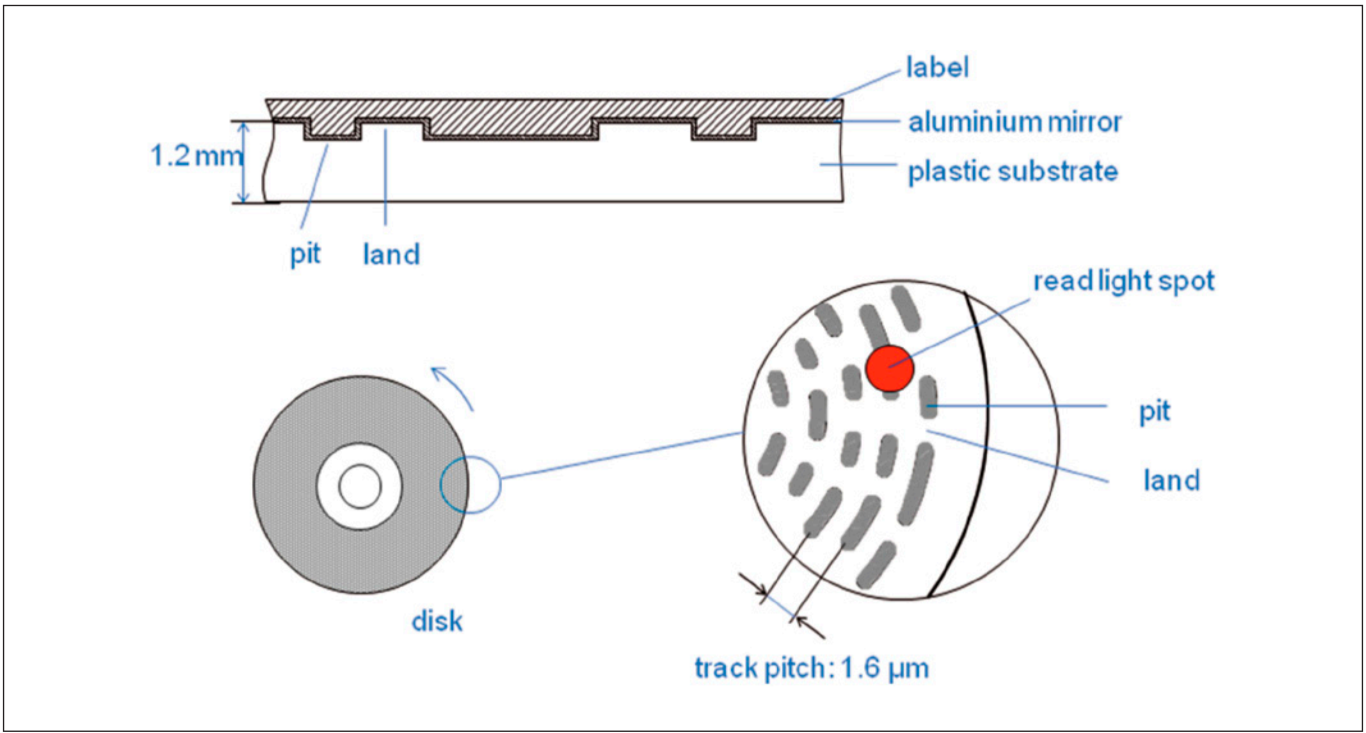


Figure 3. *Details of the CD surface structure.*

On the selection of wave length a similar problem ensued. The only diode lasers available at the time were prototype samples designed for fiber communication, operating at a wave length of 820 nm. Pressed by the need for the highest data capacity, we decided to agree on 780 nm as the wave length for CD, a decision that was based on a very limited number of 780 nm laser prototypes. During our joint experiments on the bits/mm² attainable, lots of test discs with different land/pit sequences were exchanged. In a quick succession of meetings between the Philips and Sony engineers, we improved each time on our previous results in information density and detection robustness. In the end, we concluded that for maximum information density, pits and lands had to have a minimum length of 3 units with one unit as quantization size. We also found that it was absolutely necessary to choose a “channel code” where pits and lands were evenly distributed over the disc. Only in that case, we would be able to filter away the effects of severe scratches and fingerprints on the disc.

In a similar way we confronted each other with different Error Correction schemes. Error correction is an integral part of any digital storage of communication system. Usually, a trade-off must be made between three parameters: correction performance, the relative number of parity bits, and cost. In line with our different initial target markets, also here Sony and Philips had different priorities. Sony was pushing for the most powerful scheme, while Philips was very critical on cost. Depending on the algorithm, a balance had to be struck between correction capability for large defects — like scratches and fingerprints— and small ones, such as random errors like noise. Because the two companies used discs and players that were still very much under development, we had to assess what level of quality could be achieved later in mass production. A lot of guess work! In the end we agreed on a scheme that combined good correction of large disc errors with a limited RAM size, using a structure with continuous data input and output, instead of the conventional block structure. This is a solution that works well for a continuous data stream like Audio, practical at a time when memory space was scarce.

3. FIXING SIZES

I remember tough discussions taking place between our two companies on the parameters for the quantization level of the audio samples. Philips’ priority was a small disc, fitting not only in Hifi decks, but also in mobile applications like the automobile and the so-called ghetto blasters. These markets were much larger than the stationary Hifi market. With a small disc of about half an hour playing time in mind, Philips wanted to limit the number of bits per sample to 14 bits. Sony was adamant in achieving perfect Audio Quality. I remember that in one of our meetings Toshi Doi tried to convince us of the necessity to choose 16 bit quantization. He did so by recording soft triangle music at both 14 bit and at 16 bit resolution: we were supposed to hear the difference. Frankly speaking, none of us heard any difference at all, but Sony's message was loud and clear: the new format must not compromise on sound quality!

We had settled on a 11.5 cm diameter disc, until Sony's president, Mr Ohga, —a former opera singer— put a new requirement on the table: the playing time had to be 74 minutes. This new requirement came out of the blue for all of us. In earlier meetings, we had worked on the basis of a maximum playing time requirement of one hour. What we did not know was that Ohga had promised his conductor friend Herbert van Karajan that his version of Beethoven's Ninth Symphony would fit on one CD only. No discussion was possible: at the highest level, a promise is a promise. One also has to bear in mind that, after Hiroshima and Nagasaki, the 9th Symphony choral "*Alle Menschen werden Brüder*" had become an important musical symbol in Japan.

Polygram, the music company of Philips, was shocked by this new requirement. Contrary to Ohga, Polygram wanted a rather limited playing time. The argument was that no music group would be able to regularly produce albums with playing time much in excess of existing LP's 40 minute playing time. Eventually, a political compromise was reached at an actual playing time of 74 minutes on the disc, while the Red Book Standard defined a playing time of 60 minutes only. Not often a new format formally denied its higher storage performance!

An unfortunate consequence of the playing time requirement was the disc size: it grew from 11.5 cm to 12 cm diameter. This last-minute change caused real problems. With great difficulty, I had convinced our media engineers to accept the critical specification on disc flatness based on an 11.5 cm disc size. I had no choice, so I told them they had no choice. In hindsight, I think we were too conservative, we could have maintained the 11.5 cm disc size, whilst squeezing the data density on the disc a bit more. However, at the time, only one trial run of one thousand discs from one "stamper" was available, so we had no real information on mass production quality of discs.

This history may sound as work only; in actual fact, we enjoyed the drinking parties in the evenings just as well. Yet, we worked very hard indeed: many times the target for next round of testing was felt as impossible to meet. And often, when proudly showing our hottest Philips data, it turned out that also Sony had met the impossible dead line, sometimes during the night before: it was a neck-and-neck race.

One of the last discussion points was the center-hole diameter. This time, we agreed within a few minutes. Our project leader, Joop Sinjou, put a tiny Dutch 10 cents coin on the table and said: why not? Indeed: why not, so we agreed on a center hole of a Dutch 10 cents coin, *i.e.*, 15 mm diameter (Fig. 4).

In June 1980, the two development teams agreed on the specification of Compact Disc. Fig. 5 shows the two teams at that decisive meeting.

The history of CD did not end by finalizing the format: actually that is where its history started. Therefore, I want to share also some of my memories on the product introduction of the CD.

Figure 4. *The center hole in the CD: a Dutch 10 cents coin.*

Figure 5. *The Philips and Sony teams at the final, decisive meeting (June 1980). The author is at the far left.*

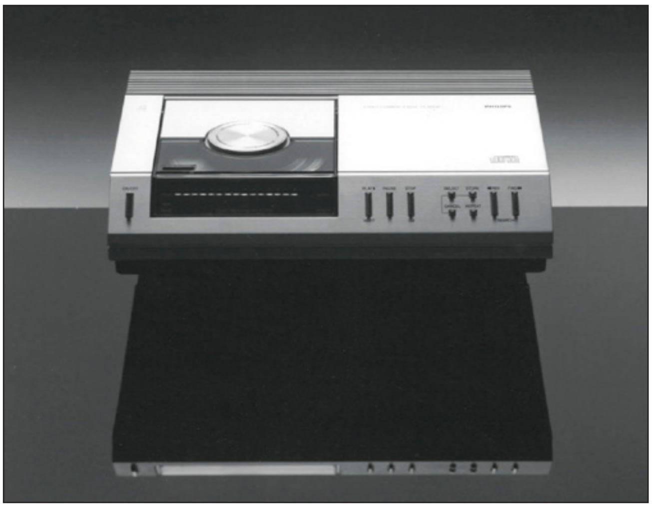


Figure 6. *The first commercial Philips CD player.*

A highlight for me was the Tokyo Audio Fair of 1982. Suddenly all Consumer Electronics companies presented their first CD players. During that Fair, I met many product engineers. They were all very proud to show the first CD player of their company. Their enthusiasm created a great feeling in all of us being part of a major innovation. And certainly, as a Philips engineer, I could be proud of our first CD player: the smallest one at the show! And with very simple and logical control and display. In one word: a beauty (Fig. 6).

When I saw the first CD player in the shop at my own little home town I felt proud. I remember what I said, with some exaggeration, to my 11-year-old daughter: Look, this is what daddy invented! But she was not impressed, which taught me another lesson: There is more to life than the CD.

About the Author

After earning his PhD in Physics from Leiden University in 1973, Jacques Heemskerk joined Philips Research in Eindhoven, where he worked on various optical problems. When, in 1979, a development laboratory was established for the Compact Disc, he joined as optical group leader. Later he became head of the laboratory, head of the optical predevelopment of Philips Consumer Electronics, and responsible for the physical part of the standardization of DVD and Blu-ray. He received the Japan Audio Society award for his contribution to CD development, the Nakajima award for CD-R, and he has been honored with a Knighthood in the '*Orde van de Nederlandse Leeuw*' for his contributions to industry and optical disc technology. He holds more than 180 patents for over 50 inventions.

Acoustics in Practice

The members of the initial editorial board are shown in the table below. New members will be added to the board to provide expertise in topic areas not sufficiently represented by the current members of the board. It is also hoped to widen the geographical spread of the board to include members form more EAA societies.

Editorial board

Name	Country	Topic
Colin English (Editor in Chief)	UK	Environmental noise and nuisance
Miguel Ausejo	Spain	Noise mapping
Arild Brekke	Norway	Building acoustics and railways
Jean-Pierre Clairbois	Belgium	Noise barriers
Victor Desarnaulds	Switzerland	Building and room acoustics
Klaus Genuit	Germany	Vehicle design
Laurent Gagliardini	France	Automotive vehicle NVH
Bart Ingelaere	Belgium	Standards for buildings
Janusz Kompala	Poland	Noise and vibration control
Maria Machimbarrena	Spain	Standards for buildings
Andrew McKenzie	UK	Environmental noise: Wind farms
Henrik Moller	Finland	Auditorium acoustics
Tønnes A. Ognedal	Norway	Offshore oil/HSE
Alexander Peiffer	Germany	Aircraft design
Patrick Van de Ponseele	Belgium	Product design and testing
Søren Rasmussen	Denmark	Environmental noise
Monika Rychtarikova	Slovakia	Building acoustics

European Acoustics Association – EAA

Comprising 32 national acoustical associations:

• Austria (AAA) • Belgium (ABAV) • Bulgaria (NSA) • Croatia (HAD) • Czech Republic (CAS) • Denmark (DAS) • Finland (ASF) • France (SFA) • FYROM (MAA) • Germany (DEGA) • Greece (HELINA) • Hungary (OPAKFI) • Iceland (IAA) • Italy (AIA) • Latvia (LAA) • Lithuania (LAS) • Morocco (MSA) • Norway (NAS) • Poland (PTA) • Portugal (SPA) • Romania (SRA) • Russia (PAO) • Serbia (ASY) • Slovakia (SKAS) • Slovenia (SDA) • Spain (SEA) • Sweden (SAS) • Switzerland (SGA-SSA) • The Netherlands (NAG) • Turkey (TAS) • Ukraine (UGA) • United Kingdom (IoA)

Serving more than 8500 individual members in Europe and beyond

The European Acoustics Association (EAA) is a non-profit entity established in 1992 that includes in its membership national acoustical societies interested in to promote development and progress of acoustics in its different aspects, its technologies and applications. The main objectives of the EAA are to:

- promote and spread the science of acoustics, its technologies and applications, throughout Europe and the entire world
- interface with associations whose activities are related to acoustics
- establish contacts across member associations and other public and private bodies
- promote the formation of national acoustical societies in European countries where these do not exist, and to support and strengthen activities of existing national associations, respecting the principle of subsidiarity
- publish a European journal on acoustics, in printed as well as in electronic format
- organize and promote congresses, publish books and monographs, and engage in all those activities that are connected with the diffusion, promotion and development of acoustics
- establish agreements for collaboration with European and international entities in order to better serve the objectives of EAA
- stimulate education activities and platforms in acoustics at all educational levels, both academic and professional
- promote and divulge the establishment and implementation of norms and recommendations in the various fields of acoustics

EAA is democratically organized (one vote per country) with a general assembly, a board and an executive council.

EAA web

www.euracoustics.org

EAA contact (General Secretary)

secretary@european-acoustics.net

EAA Board 2013-2016

President: *Michael Taroudakis*
Vice President: *Jean Kergomard*
Vice President: *Mats Åbom*
General Secretary: *Tapio Lokki*
Treasurer: *J. Salvador Santiago*

EAA Office

Antonio Perez-Lopez
c/o: Spanish Acoustical Society (SEA)
Serrano 144, ES-28006 Madrid, Spain
office@european-acoustics.net

Technical Committees

EAA has 7 technical committees which, at different level, are in charge of organizing specific activities (technical reports, round robin tests, structured session organization at congresses, symposia, etc.). They are open to all individual members of EAA member societies and are coordinated by a Chairman:

• CA, Computational Acoustics • HYD, Hydroacoustics • MUS, Musical Acoustics • NOI, Noise • PPA, Psychological and Physiological Acoustics • RBA, Room and Building Acoustics • ULT, Ultrasound

EAA is an Affiliate Member of the International Commission for Acoustics (ICA)

INTERNATIONAL COMMISSION FOR ACOUSTICS ICA

and Member of the Initiative of Science in Europe (ISE)

Initiative for Science in Europe ise

EAA Products

ACTA ACUSTICA united with ACUSTICA
Product Manager and Editor in chief: *Jean Kergomard*
Acta Acustica united with Acustica is an international, peer-reviewed journal on acoustics. It is the journal of the

EAA. It is published by S. Hirzel Verlag • Stuttgart. See www.acta-acustica-united-with-acustica.com for more information.
EAA members receive Acta Acustica united with Acustica online as part of their membership.

NUNTIUS ACUSTICUS-EAA NEWSLETTER
Product Manager: *Kristian Jambrošić*
Nuntius Acusticus is the "acoustic messenger" of EAA to vitalize communication between and in the European acoustical societies on a variety of topics. It is published monthly in electronic format and distributed via e-mail to all EAA members.

DOCUMENTA ACUSTICA
Product Manager: *Sergio Luzzi*
Documenta Acustica is the literature distribution system of the EAA. It distributes conference and symposia proceedings as well as books, reports and theses.

FENESTRA
Product Manager: *Olivier Dazel*
Fenestra Acustica is the website of EAA. Fenestra provides information on the association and its members (products, technical committees, organisational structure and policies, contact information), up-to-date news, upcoming events, links to other no-profit organisations in acoustics, a job market and much more.

SCHOLA
Product manager: *Malte Kob*
Schola is an online platform for education in acoustics in Europe: https://www.euracoustics.org/activities/schola. Through Fenestra, it offers information on university acoustics courses in Europe at different levels (Bachelor, Master, Ph.D.).

ACOUSTICS IN PRACTICE
Product manager: *Colin English*
This new technical journal will be written by practitioners for practitioners and other professions: a new link between all members of all EAA societies. The journal will be published four times a year in electronic form only.

YOUNG ACOUSTICIANS NETWORK
Contact person: *Xavier Valero*
This network is a non-profit initiative within the European Acoustics Association (EAA) with the primary goal is to establish acommunity for young researchers and young professionals in the field of Acoustics; to connect them and to provide support. It organises events at conferences and provide services that contribute to the community, such as a monthly newsletter and many communication channels to enable networking.

FORUM ACUSTICUM
Forum Acusticum is the triennial international convention organised by a national acoustical society on behalf of EAA. It is, in effect, a forum comprising a variety of different activities: high-quality scientific congress with invited plenary lectures, structured sessions, invited and contributed papers, an exhibition that includes commercial firms, laboratories and agencies, social meetings of acousticians with receptions, visits and awards.

EURONOISE
Euronoise is the European Conference and Exhibition on Noise Control, coordinated by the EAA Technical Committee Noise and organised by a national acoustical society on behalf of EAA.

EUROREGIO
Euroregio is an expression of EAA support for traditional regional events organized by groups of countries. Where and when appropriate, the regional events can be extended towards a full European and international scale.

EAA SYMPOSIA
EAA symposia are scientific meetings under the aegis of the EAA with a focus on specialised fields. They are typically organized by one or more member societies of EAA in conjunction with the Technical Committee of EAA.

YOUNG RESEARCHER AND STUDENT PROGRAM
EAA supports with grants and best paper and presentation awards the active participation of students and young researchers at EAA major events (Forum Acusticum, Euronoise, Euroregio).

EAA SUMMER AND WINTER SCHOOLS
The EAA Summer and Winter Schools are conceptualized as events where Master and PhD students of acoustics, as well as other young acousticians, can learn about a variety of new accomplishments in the field of acoustics in half day or full day courses.

European Acoustics Association (EAA)
secretary@european-acoustics.net
office@european-acoustics.net

www.euracoustics.org